VoIP with QoS and Bandwidth-on-Demand for DVB-RCS

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Abstract

This paper proposes a consolidated approach for Voice over IP (VoIP) with Bandwidth on Demand over satellite networks based on the ETSI DVB-RCS standard. A real-time service like voice communication needs priority over other services in IP environments with limited bandwidth. In satellite networks bandwidth utilization should be optimized in order to save service costs, which requires dynamic bandwidth allocation schemes, and we study trade-off between voice quality and bandwidth efficiency under different DVB-RCS-specific capacity request and allocation strategies. It is demonstrated that DVB-RCS provides an efficient platform for integrated support for a variety of VoIP applications over satellite. The main contribution of this paper consists in the identification of the mechanisms capable of responding to the key challenges raised by the VoIP application in satellite environment.

1. Introduction

DVB-RCS [2] is the open ETSI standard for broadband satellite communications with multiple vendors. The specification targets mainly the lower layers; the higher layers and the interoperability between vendors is handled by SatLabs, an independent non-profit organization initiated by ESA. DVB-RCS has several powerful BoD mechanisms that can be used for providing prominent applications such as VoIP with adequate Quality of Service (QoS), but the interaction between the IP layer, where QoS is set, and the lower layers, where the traffic is finally prioritized for transmission, is not covered by any specification. A purpose of this paper is to define and demonstrate a consolidated interoperable approach for this interaction.

VoIP is rapidly gaining popularity. An important reason for this is that VoIP offers low cost telephony to users and can easily and incrementally add value, e.g. integrate multimedia. For Service Providers (SPs), VoIP offers the desirable convergence between packet-switched networks and traditional circuit-switched networks. However, the Internet was originally developed to provide best effort (BE) services for data applications generating bursty traffic. Its use for real-time applications such as VoIP raises significant challenges over bandwidth-constrained satellite links, and delivering VoIP over satellite can be seen as a QoS and resource management problem.

QoS management relies on network capabilities at session, connection and transport layers [1]. VoIP uses the Real-time Transport Protocol (RTP, RFC3550) for the delivery of voice packets over UDP/IP, and relies on forwarding mechanisms at the IP and MAC layers. IP-QoS mechanisms are based on the IntServ or DiffServ frameworks, while MAC mechanisms are network-specific.

This paper is organized as follows: Section 2 provides an overview of VoIP specifics and QoS provisioning. Section 3 describes a generic architecture for VoIP over satellite. Section 4 investigates the use of BoD in DVB-RCS networks for supporting VoIP. Section 5 includes discussion and conclusions.

2. VoIP Background

2.1 Reference Architecture

In support of VoIP and QoS provisioning over DVB-RCS, SatLabs[1] is defining standard architectures and mapping mechanisms of IP layer QoS into the DVB-RCS MAC layer.
The simple SatLabs-based reference architecture in **Error! Reference source not found.** for VoIP over satellite includes the satellite terminal, VoIP ATA (Analog Terminal Adapter) and customer LAN on one side of the satellite, and the gateway with standard VoIP equipment for connection to PSTN and Internet on the other side of the satellite. The customer LAN may include an IP telephony server (e.g. IP PBX), analog phones and adapter, IP phones or soft-phones.

2.2 Performance Metric

The following key VoIP performance parameters are relevant:

- Delay, the one-way delay experienced by packets between source and destination. For satellite network it is dominated by the propagation delay (typ. 270 ms for GEO satellites).
- Jitter, defined as variation of the packet delay. Jitter may lead to voice distortions due to e.g. too small buffers, lack of adaptive behaviour or sudden increases in delay.

[1] SatLabs members are committed to bringing the deployment of the open, multi-vendor DVB-RCS based systems to large-scale adoption. SatLabs complements DVB-RCS with recommendations and guidelines for equipment interoperability and provides a mechanism for formal DVB-RCS interoperability certification.

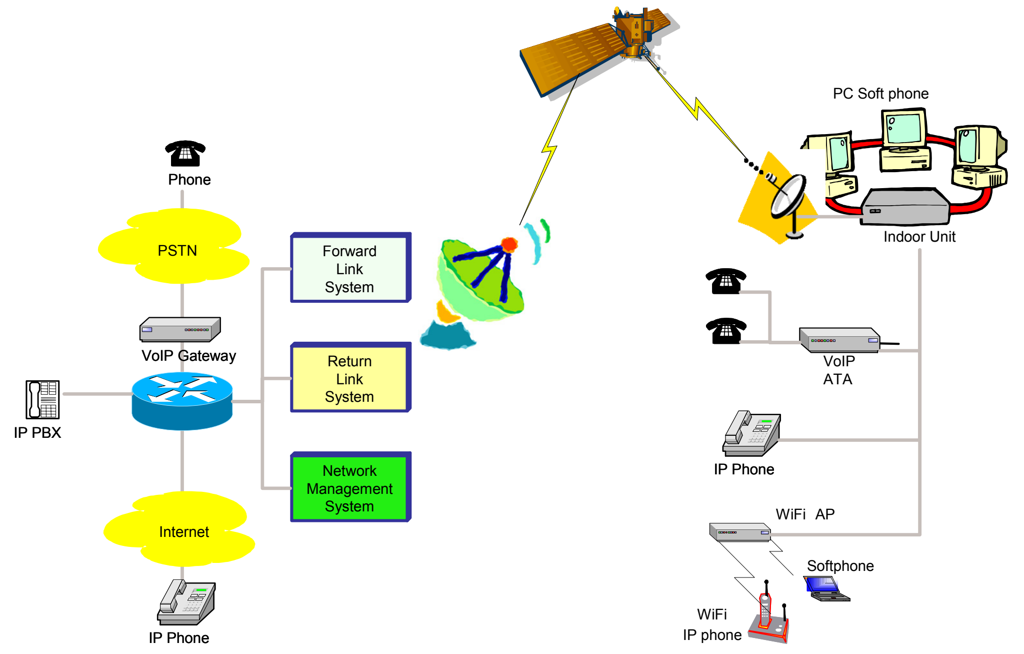

Figure 1. Reference VoIP over Satellite Architecture

- Packet loss, the result of propagation delays and channel impairments (noise, interferences). It can cause audible gaps in the speech sequence.
- Bandwidth, for compressed voice depends on the voice codec. Bandwidth efficiency is associated with complexity, algorithmic and processing delay.

The first three parameters above are QoS parameters, while the last one is associated with network resource utilization. One must jointly optimize all these parameters based on trade-offs. For instance, larger buffers would help reduce the jitter but add more delay. Similarly, improving bandwidth efficiency by placing several voice frames into one IP packet would also add more delay.

Two important but contradictory aspects in delivering VoIP are the bandwidth commitments and delivery time; static capacity allocation and over-provisioning provides good delivery time but poor bandwidth utilization. A better option is to apply dynamic capacity allocation, as described in this paper.

2.3 VoIP bandwidth

The voice coders G.729 and G.723 are commonly used with VoIP, and compress the voice down to 8 kbps or 6.3(5.3) kbps, respectively, with close to toll quality. Voice coding data is encoded and output in blocks, typically of 10 or 20 ms duration. More blocks packed into an IP packet result in the less IP overhead at the cost of additional delay corresponding to the number of blocks packed. Taking into account RTP/UDP/IP header, a G.729 stream @ 8 kbps will require 16 or 24 kbps capacity at the IP level with 4 or 2 voice packets per IP packet, respectively.

Voice in 2-party conversations has a typical activity factor of 40%, and during silence periods the codec encodes just background noise. It is then possible to detect silent frames and suppress the coding or use a significantly lower data-rate, giving a bursty source. Modern coding techniques both for voice, audio and video commonly use Variable Bit-rate (VBR) coding. In cases where the bit-rate is not constant, a resource management system must be able to allocate required bandwidth dynamically in order for users and the network to benefit

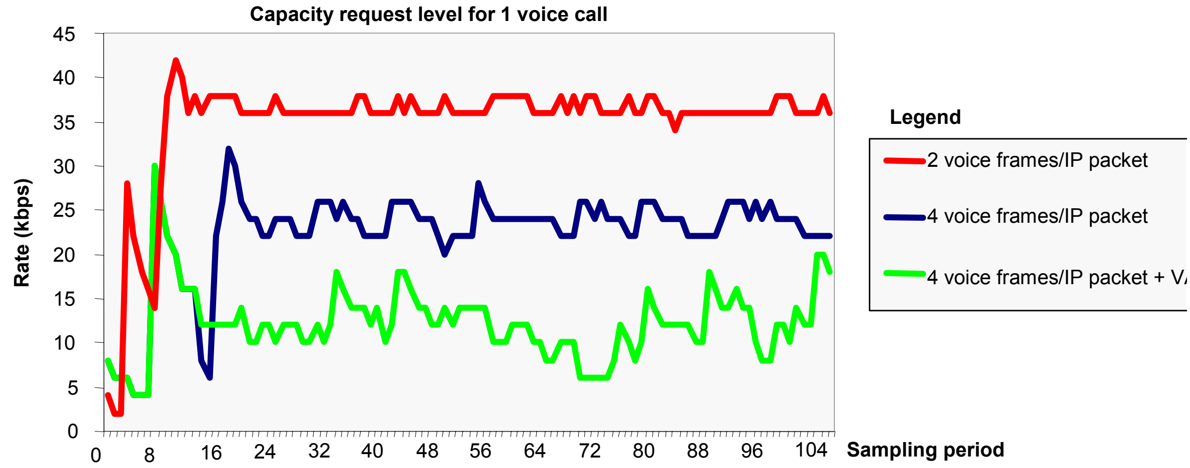

Figure 2: Performance of VoIP showing requested BoD for three scenarios.

from the lower capacity requirements. If there are more sources e.g. many speakers, then the bit-rate variance is reduced, and from around 20 sources it could be well approximated by a constant bit-rate source. Bandwidth requests for two different packetizations (2 and 4 voice frames per IP packet), as well a case with VAD are illustrated in Figure 2.

3. QoS provisioning

VoIP using BoD satellite networks requires the implementation of QoS mechanisms at various network protocol layers. A survey of QoS techniques for satellite networks is found in [4]. Two IP-layer QoS frameworks are defined;

- Integrated Services (IntServ) and
- Differentiated Services (DiffServ).

Both define traffic forwarding rules for different classes of service. For IntServ this is done per flow with an end-to-end point of view, requiring end-to-end signaling (i.e. RSVP) for resource reservation along the path. IntServ can provide end-to-end service guarantee, but raises scalability issues since per-flow states need to be maintained in all network nodes (routers) along the end-to-end path. We will not study IntServ further here.

DiffServ resolves the scalability problems by replacing the per-flow service with a Per-Hop Behavior (PHB), applicable to traffic aggregates, so that per-flow states in network nodes are no longer required. End-to-end performance is then provided by the concatenation of multiple PHBs. Five PHB classes have been defined by DiffServ (RFC2475), namely Expedite Forwarding (EF) and 4 four Assured Forwarding (AF) classes, in addition to the Best Effort (BE) service of the original Internet protocol. At each network node each packet is treated according to its class, as defined by the Type of Service (ToS) bits in the packet header. The treatment is performed by the DiffServ mechanisms and includes traffic classification, conditioning (metering, shaping, dropping, marking), queuing and scheduling. DiffServ mechanisms need to be configured with QoS-related parameters and policies, either statically or dynamically.

DiffServ is the model of choice for most satellite systems. VoIP packets will be mapped into a PHB class that provides rate guarantee (e.g. EF), so they will be transmitted before data associated with a lower priority service class (e.g. AF or BE). DiffServ offers a relative QoS, but if the loading in each class (compared to the configured bandwidth) is controlled, an implicit average performance is guaranteed to all flows in the class. This is particularly relevant to applications requiring QoS guarantees, such as VoIP.

Packet forwarding at IP layer depends on services offered by the MAC layer. In the case of BoD satellite systems the MAC layer techniques play an important role with regard to the return link resource control. On the forward link the MAC support is much simpler and consists primarily in configuring "virtual pipes" with appropriate bandwidth.
In order to provide end-to-end QoS guarantees, DiffServ nodes along the path need to be configured with adequate bandwidth using session / connection control layer signaling.

- Session layer signaling is used to establish application sessions (e.g. VoIP) between end users with pre-defined or negotiated session parameters.
- Connection control layer signaling is used to set-up the transport network (e.g. satellite network) according to the session parameters for each media component (e.g. VoIP).

4. Bandwidth on Demand

4.1 BoD in DVB-RCS

The MF-TDMA time slots in DVB-RCS are dynamically allocated to terminals based on demands (commensurate with the transmit traffic) and resource availability. Some slots can also be statically allocated to certain terminals. The allocation (static or dynamic) is performed by the BoD scheduler, which implements the MAC protocol/algorithms. The allocations are regularly broadcast over the forward channel via Terminal Burst Time Plan (TBTP) messages. Upon receiving the TBTP, the terminals transmit their traffic in the allocated timeslots. The terminals are responsible for calculating the capacity requests and for dispatching the traffic from various queues associated with the operation of the BoD scheduler.

The basic problem at the physical layer is simple: What data should be transmitted next. However, the solution is non-trivial in a mixed traffic case. IP classes of service need to be mapped into equivalent MAC QoS classes where the priorities established at the IP layer are preserved at the MAC layer via a multiple-queuing strategy. Various algorithms calculate the capacity required for each queue. Given the propagation delay and traffic variability, prediction methods may be used to anticipate queue growth and adjust the requests. The multiple queue strategy needs a scheduling mechanism that prioritizes the voice packets in order to reduce the queuing delay. Priority scheduling also ensures that VoIP performance is not affected by the presence of other traffic.

Large delays have a negative impact on BoD-schemes. The access delay varies according to the monitoring/allocation interval, capacity request strategy, request signaling strategy and bandwidth allocation scheme. The monitoring interval should be short enough to capture traffic variations but long enough to prevent excessive request overhead. The allocation interval (or time slot assignment periodicity) is usually equal to the length of the superframe. With a superframe of 125ms, the voice is sampled 8 times per second - enough to ensure good quality. The signaling strategy should reduce the system access delay as much as possible.

4.2 Bandwidth allocation schemes

In a DVB-RCS satellite network the return resources are subject to contention among terminals. The contention is resolved by using a protocol from the CF-DAMA (Combined Free - Demand Assignment Multiple Access) family for capacity assignment. The protocol relies on several distinct capacity categories, i.e.

- Constant Rate Assignment (CRA),
- Rate Based Dynamic Capacity (RBDC),
- Volume Based Dynamic Capacity (VBDC) and
- Free Capacity Assignment (FCA).

While CRA and RBDC assignments are guaranteed, VBDC and FCA assignments are of best effort nature. RBDC and VBDC are dynamically assigned. The capacity categories, used alone or in combination, form the basis of different bandwidth allocation schemes, or strategies, capable of providing service differentiation, QoS guarantee and bandwidth utilization at different granularity levels.

- Fixed-rate assignment is a guaranteed assignment scheme providing CRA for the duration of terminal logon period or on per-time basis. Capacity is allocated whether the terminal has traffic to transmit or not, but all traffic may share the capacity.
- Fixed-rate BoD assignment is a guaranteed assignment made available when the terminal needs it (e.g. for the duration of the VoIP session), therefore it is referred to as "dynamic" CRA and is made available with the latency of session/connection establishment. If not used by the intended application it may be used by other applications.
- Variable-rate BoD assignment is a dynamic assignment scheme where a variable number of time slots are assigned based on explicit requests. When based on RBDC, traffic slots are assigned each superframe in order to sustain the requested rate, until a new request is made or until a time-out expires; capacity is guaranteed up to a limit value (MaxRBDC) if overbooking is not allowed. When based on VBDC, traffic slots are typically assigned as available to a cumulated volume, based on some fairness criteria (e.g. round-robin). VBDC fits well to the typically bursty Internet data traffic.
- The Free assignment scheme distributes capacity - that would not be used otherwise - across terminal population, thus improving the latency performance. No signalling from terminals is involved. If there is spare capacity in a network free capacity may be allocated without it being requested.

Connections support transfer capabilities based on three types of bandwidth parameters:

- Guaranteed Constant Rate: granted for permanent connections according to some service level agreements. This is supported by CRA with no signalling involved.
- Non-guaranteed Constant Rate: granted or refused for on-demand connections at setup. This is supported by CRA, involving signalling between terminal and hub.
- Non-guaranteed Variable Rate: granted based on capacity requests. This is supported by RBDC and/or VBDC.

The allocation schemes are optimized with respect to different criteria, like latency, jitter or capacity utilization. The first two of particular relevance for VoIP traffic delivery; low latency means that capacity is made available almost instantly, while low jitter means that the time slots are evenly assigned. Fixed-rate schemes are optimized for both delay and jitter, but offer low scalability; even a small amount of CRA configured for each terminal would severely limit the number of active terminals. A fixed-rate BoD scheme, also jitter free, offers better efficiency and good responsiveness. Variable-rate BoD assignment is optimized for efficiency; the access delay can be low, depending on the capacity request and signalling strategy. Free assignment schemes are inappropriate for VoIP service. Allocation schemes may be combined in order to guarantee the required QoS associated with specific VoIP services.

4.3 Capacity Request Strategies

Different capacity request techniques support different traffic types. In the context of mixed traffic the overall BoD picture becomes complex, requiring advanced strategies for capacity calculation. The CR strategies allow the DVB-RCS capacity categories to be efficiently combined in order to maximize the QoS and/or return link efficiency, by mapping the QoS traffic classes to capacity categories. An example of mapping is illustrated in **Error! Reference source not found.**. Given a set of user-defined configurable QoS classes {RT, VR, JT}, a set of pre-defined CR strategies {RB_LG, RB_HG, RB_MG, VB_UTIL, VB_PERF, VoIP_PEAK} and the set of DVB-RCS capacity categories {CRA, RBDC, VBDC, FCA}, the following DiffServ mapping can be defined:

QoS Class -> Capacity Request Strategy -> Capacity Category

It allows various mapping of DiffServ PHBs into the DVB-RCS capacity categories. As the names suggest ("HG" High Grade; "MG" Medium Grade, "LG" Low Grade, "RB" Rate-based, "VB" Volume-based, "PERF" performance; "UTIL" utilization), the CR strategies implement different performance-utilization trade-offs, by combining rate- and volume-based techniques. The set of QoS traffic classes ("RT" Real Time, "VR" Variable Rate and "JT" Jitter Tolerant) recommended in [6] correspond to the DiffServ EF, AF and BE PHBs, respectively. The voice traffic would be mapped to the RT traffic class corresponding to EF PHB. An example of mapping in the context of ATM classes can be found in [7].

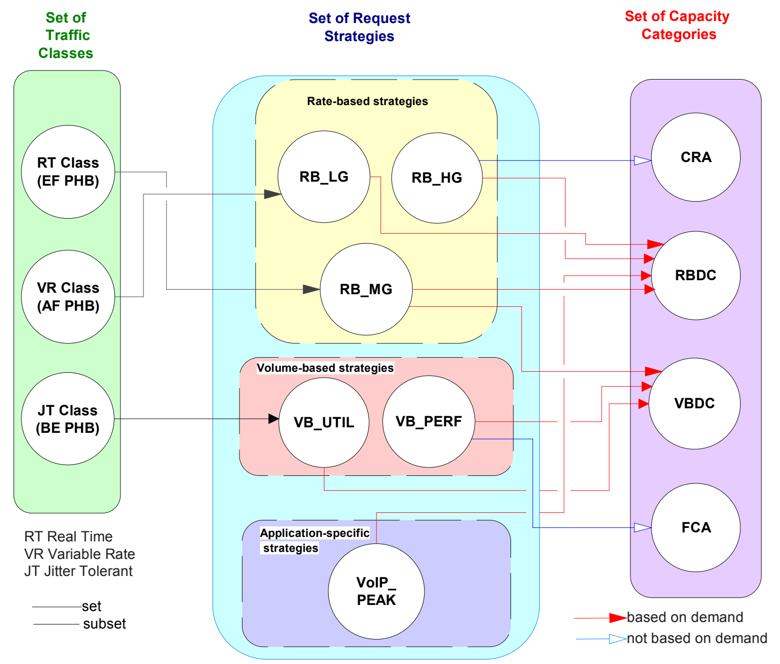

Figure 3: Example of DiffServ mapping to capacity categories

The right CR strategy for VoIP service ultimately depends on what the service is needed for. For a quality-critical application, the RB_HG would be the most appropriate CR strategy, trading bandwidth for performance and delivering high quality voice regardless of the congestion level. Non-critical VoIP can be mapped to a cost-aware RB_MG strategy, trading delay for utilization and lower cost. A CR strategy, combining RBDC and VBDC for an acceptable trade-off between service quality and cost, is presented in [[8]]. While RBDC provides sustained assignments during VoIP activity periods, VBDC is used for BE traffic and to maintain ("keep-alive") request opportunities when there is no traffic, in order to improve the transitions from silence to activity.

5. Discussion and Conclusion

While QoS performance parameters are important for VoIP, a particularly relevant aspect is how BoD capacity is made available. Static resource allocation for VoIP is not optimum in a satellite environment but high-quality voice communications can be achieved with dynamic capacity allocation, based on standard MAC layer mechanisms.

Better capacity utilization is achieved through technical advances in header and voice compression, packetization and silence suppression. On satellite links the delay is more and more frequently traded for bandwidth, by packetizing more voice frames per IP packet order to reduce the capacity needs. With G.729 and 4 voice frames per IP packet for example, the MPEG encapsulation efficiency is 83,8% and the theoretically required MPEG rate becomes 19 kbps. The real world values will depend on the efficiency of the applied BoD scheme. Along the same lines, silence suppression contributes significantly to increased utilization in the context of mixed traffic. It is important to keep in mind that performance and efficiency are in general two conflicting goals and each network needs to investigate its own boundaries in order to choose the appropriate scheme and dimension the capacity accordingly.

The following issues should be considered for allocating BoD for VoIP :

- Identify proper traffic profile
- Minimize access delay
- Enable QoS
- Minimize jitter
- Smooth bandwidth allocation

The best solution is to optimize the BoD schemes for mixed traffic in order to obtain maximum capacity utilization while guaranteeing QoS for all applications. With several speakers and mixed traffic, the capacity allocation scheme is simplified.
In conclusion, the DVB-RCS is now a mature technology, with suitable QoS methods, policies and protocols, which can efficiently deliver VoIP services. By adopting the right strategies in the allocation of capacity, DVB-RCS provides a suitable, yet cost-efficient platform to support VoIP and in general applications/services requiring QoS support.

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