Planning of VoIP System for Hybrid Fiber-Coaxial Television Networks

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***Abstract* – Introducing VoIP into cable television network is an efficient and inexpensive alternative of the traditional switched telephone network. It is assumed that the cable television network corresponds with the EuroDOCSIS standards. When a VoIP system is planned, the cable network capacity and architecture, required number of simultaneously performed calls, and number of allowed call attempts per second are taken into consideration. VoIP system capacity is estimated depending on network loading and used codec for transmission channel parameters defined in the DOCSIS specifications.**

***Keywords* – Voice over IP, DOCSIS, HFC Network, CATV**

I. Introduction

Hybrid fiber-coaxial (HFC) cable television network is a wide used urban access network that offers large variety of interactive services [1, 2]. For voice services delivering in this type of networks voice over IP (VoIP) technology is the most used.

VoIP service operates by converting voice signals to data packets, sending these data packets through the IP network, converting these packets back into voice signals, and managing the overall call setup (dialing), connection, and termination (hang-up) [3].

VoIP introduces new traffic loads and traffic that is synchronous in nature, versus asynchronous like typical residential high speed data. VoIP also requires support for low delay and low packet loss. This is because VoIP is a streaming service where retransmission is not feasible. In addition VoIP requires high service availability. The level of availability depends on whether the VoIP service is intended to be a primary line service, available after a power outage or a secondary line service where service availability is not as critical.

For just the CMTS and HFC portions of a cable operator's network, VoIP design choices span:

– Multimedia Terminal Adaptor (MTA) configuration parameters;
– HFC node and DOCSIS parameters;
– CMTS configuration parameters.

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II. Architecture of HFC Cable Television Network with VoIP Service Delivering

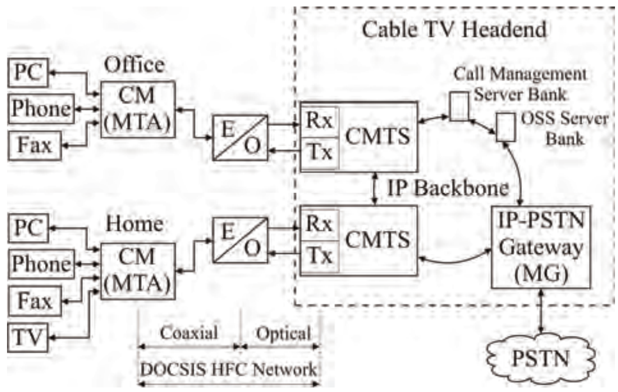

Fig. 1. Architecture of HFC cable TV network with VoIP service delivering

IP telephony equipment can be easily included into existing cable architecture (Fig. 1) [4, 5]. Cable modem termination system (CMTS) is an essential network component that enables voice packet transmission over HFC network. CMTS consists of upstream RF receivers and a downstream RF transmitter. It is connected by optical links to HFC access network, and to three large groups of servers. The first group is responsible for the operation system support (OSS). The second group is a call management server, and the third group contains IP-PSTN Gateways. The OSS server group is responsible for installation, configuration, operation, and management of the entire system. The call management server group tends on the signaling information condition and the connection control. For connectivity with the public switched telephone network PSTN-gateways are needed [6].

Described components correspond the PacketCable specifications for voice over cable television network [7].

III. Planning of VoIP System for Cable TV Network

A. Multimedia Terminal Adaptor VoIP configuration parameters

There are many MTA configuration parameters but those with the largest impact on VoIP service include VoIP codec, packetization period, voice activity detection (VAD) state, and jitter-buffer size. The VoIP codec defines the method by which voice is encoded and is often referenced in terms of its output bit rate. The VoIP packetization period is the period over which encoded voice bits are collected for encapsulation in packets. The VAD state refers to whether VoIP packets are sent all the time or only during talk periods, with VoIP packets not being transmitted during quite periods. The VoIP jitter-buffer size defines the maximum delay and nominal play-out delay of a jitter buffer.

The choice of the MTA configuration parameters impacts capital cost from a CMTS utilization perspective and operational cost from a VoIP QoS perspective. Basic MTA configuration parameters are shown in Table I.

TABLE I
MTA CONFIGURATION PARAMETERS

Configuration parameters	Typical configuration today
Codec	G.711, but G.728 and G.729E are optional recommendations iLBC and BV16 are mandated in PC1.1
Packetization period	10ms or 20ms (20ms and 30ms only options for iLBC)
VAD state	Avoided in cable applications
Jitter buffer size	Variable, but not uncommon to see 15ms play out and 30ms maximum

A lower rate codec will reduce VoIP payloads allowing higher channel utilization, but will also results in lower voice quality, and may degrade FAX and modem performance and prevent the passing of inband DTMF tones. A lower rate codec also imposes a higher processing burden on the MTA as well as the multimedia gateway (MG). This may result in MTA call-mixing limitations and lower MG capacity. Most cable VoIP trials and early deployments today use G.711 encoding with its *64kbps* pulse code modulated output. But PacketCable, the cable VoIP standards forum, has several low-rate codec recommendations and requirements that are categorized as providing a “toll-grade” voice performance with much lower output rates [7].

Table II provides an example of the reduced-rate benefits of low rate encoding.

TABLE II
EXAMPLE OF CODEC IMPACT ON CALL RATE

Codec	Codec output rate	QPSK upstream call flow rate	Downstream call flow rate
G.711	64 kbps	115.2 kbps	109.6 kbps
G.728	16 kbps	57.6 kbps	61.6 kbps
G.729E	12 kbps	57.6 kbps	57.6 kbps

It shows raw output rates and DOCSIS service flow rates (including overhead from RTP, UDP, IP, Ethernet, and DOCSIS protocols) for G.711, G.728, and G.729E encoding. The example assumes *10ms* packetization periods and use of DOCSIS PHS, BPI+, and FEC (PHS assumptions are for *41* byte savings in upstream and *12* byte savings in downstream. Upstream FEC assumptions are for long data grants with RS $K=220$, $T=8$, short data grants with RS $K=78$, $T=5$, and use of shortened last code words. In the example G.711 packets are assigned to long data grants and G.728 and G.729E packets are assigned to short data grants.).

Referring back to Table I, we next see that a higher packetization period produces less packet overhead thereby allowing higher channel utilization. However a higher packetization period also causes higher VoIP path delay and can increase packet error rate (*PER*). Packetization delay can be codec dependent and nominally contributes around *1.5* times the packetization period to VoIP path delay. The path delay contribution is the result of the packetization delay itself plus DOCSIS unsolicited grant service (UGS) grant uncertainly. UGS grant uncertainty is the time a newly generated voice packet must wait at the MTA’s CM before receiving a UGS grant transmission opportunity. Nominally this delay is half the packetization period. Because of these concerns, most cable VoIP deployments today use *10ms* or *20ms* packetization periods.

Table III provides an example of the trade-off between packetization-period delay and rate reduction. It shows nominal path delay and DOCSIS service flow rates for *10*, *20*, and *30ms* packetization periods. The example assumes G.711 encoding and use of DOCSIS PHS, BPI+, and FEC.

TABLE III
EXAMPLE OF PACKETIZATION PERIOD TRADE-OFF

Packetization period	Nominal path delay contribution	QPSK upstream call flow rate	Downstream call flow rate
10 ms	15 ms	115.2 kbps	109.6 kbps
20 ms	30 ms	89.6 kbps	86.8 kbps
30 ms	45 ms	85.3 kbps	79.2 kbps

Referring back to Table I, we next see that enabling VAD will reduce VoIP load since voice will only be sent when talking occurs. This allows more calls and/or data to exist concurrently, resulting in higher channel utilization. However enabling VAD will also degrade voice quality as the result of a voice clipping at the beginning of each talk period. All cable VoIP trials and early deployments that we know of today avoid VAD for this reason. All cable VoIP trials and early deployments that we know of today avoid VAD for this reason.

Finally from Table I, we see that MTA jitter-buffer size has no impact on CMTS utilization and should be chosen based on VoIP QoS considerations. Jitter buffer size should provide an appropriate balance between jitter-induced packet drops and its contribution to VoIP path delay. Choosing a large jitter-buffer reduces packet dropping from jitter but increases VoIP path delay. In particular, its selection should be coordinated with VoIP *PER* considerations for a total packet loss rate target, and with the choice of VoIP packetization-period to meet a VoIP path delay target.

To determine an appropriate VoIP path delay target, first consider an appropriate maximum round-trip VoIP delay. *300ms* is often used as a target for maximum round trip delay between an MTA and PSTN phone over a long distance connection. From this delay, *150ms* is often targeted for end-to-end delay, with around *100ms* assigned to PSTN long-distance propagation delay. The result in this case is a remainder of *50ms* for local VoIP path delay.

Jitter-buffer size, packetization period, and packetization-related grant uncertainty combine with other delays to form a VoIP path delay.

B. HFC node and DOCSIS parameters

HFC node parameters that impact VoIP service include average node size and the maximum number of nodes per CMTS receiver group. Average node size defines the average number of homes served by a fiber node. Larger node sizes require fewer fiber terminating nodes and fibers. Smaller nodes generally result in less ingress noise in an upstream channel.

The maximum number of nodes per CMTS receiver group defines the maximum number of nodes that can be connected to one or more CMTS receivers. Higher node allowance per receiver group permits more nodes to be supported per receiver when service take rate is low. But larger allowance will also funnel more ingress noise in an upstream channel.

The choice of these parameters impacts capital cost from a node and CMTS utilization perspective and operational cost from a VoIP QoS perspective.

TABLE IV
DOCSIS CHANNEL PARAMETERS

DOCSIS channel parameter	Typical configuration today
Upstream bandwidth	1.6 or 3.2 MHz
Upstream modulation	QPSK or 16QAM (only choices)
Downstream modulation	64QAM or 256QAM (only choices)
PHS	~40 bytes upstream, not clear for downstream
Upstream FEC	Dependent on upstream conditions

DOCSIS parameters that impact the VoIP service include upstream channel bandwidth, upstream and downstream modulation, upstream and downstream packet header suppression (PHS), and upstream forward error correction (FEC) (Table IV). EuroDOCSIS allows for a variety of upstream bandwidth and upstream and downstream modulation choices [8, 9]. Choosing *3.2MHz* upstream bandwidth provides double the capacity of *1.6MHz* channel but requires support for a contiguous chunk of *3.2MHz* bandwidth within upstream band. It also requires a *3dB* higher signal to noise ratio (*SNR*). A 16-QAM upstream modulation format provides double the channel capacity of QPSK. But 16-QAM also has approximately *7dB* higher *SNR* requirements than QPSK at 10^{-6} *BER*, and greater amplitude and phase noise sensitivity. A 256-QAM downstream modulation format provides over *40%* higher channel capacity than 64-QAM. But 256-QAM also has approximately *6dB* higher SNR requirement than 64-QAM at 10^{-6} *BER*, and significantly greater amplitude and phase noise sensitivity.

PHS defines the amount of packet overhead savings in the upstream and downstream channels accompanying VoIP packets using UGS. The PHS reduces VoIP overhead allowing higher channel utilization, and has no impact on VoIP QoS.

FEC defines Reed Solomon coding parameters used in upstream channels. Limiting the amount of FEC will help limit VoIP overhead and allow higher channel utilization. But increasing the amount of FEC improves noise and interference robustness, possibly permitting support for higher bandwidth and modulation choices.

The choice of these parameters impacts capital cost from a CMTS utilization perspective and operational cost from a VoIP QoS perspective.

C. CMTS configuration parameters

CMTS configuration parameters that impact VoIP service include the maximum bandwidth to allocate to VoIP service and the ratio of node-to-CMTS receiver assignment. The maximum bandwidth allocation for VoIP defines the maximum upstream and downstream bandwidths allowed for VoIP service at peak VoIP loading. This allocation is essential when data is sharing the same DOCSIS channel resources because it prevents VoIP calls from monopolizing all available bandwidth. The ratio of node-to-CMTS receiver assignment defines the number of nodes assigned to a CMTS receiver group and the size of the receiver group in terms of number of receivers. For example the ratio may be *1* node per receiver group of *2* receivers.

The choice of CMTS parameters impacts capital cost from a node and CMTS utilization perspective and operational cost from a CMTS and node reconfiguration and VoIP QoS perspective (Table V).

TABLE V
CMTS CONFIGURATION PARAMETERS

Configuration parameters	Typical configuration today
Maximum VoIP bandwidth allocation	Dependent on data traffic load, but typically between 40% to 60%
Ratio of node-to-CMTS Rx assignment	Unclear, but Rx group size >1 is likely to increase as VoIP take rate grows

The use of current versus padded traffic margin when choosing a maximum VoIP bandwidth allocation provides a trade off between CMTS utilization and frequent reconfiguration. In addition, a node-to-CMTS receiver assignment that allows more than 1 receiver per receiver group can avoid the need for node splitting at high service take rates.

The optimal choice of maximum VoIP bandwidth allocation and the ratio of node-to-CMTS receiver assignment requires an iterative computation to determine the maximum take rate for a given node size (or node size maximized for a given take rate). The computation requires voice and data traffic load assumptions along with traffic margin (typically added to take rate but could involve setting DOCSIS parameters, as well as CMTS hardware design and DOCSIS parameters, as well as CMTS hardware design and performance constraints.

TABLE VI
EXAMPLE OF NUMBER OF LINES PER CMTS ESTIMATION

Used codec	Packetization Period, ms	QPSK upstream call flow rate, kbps	Number of simultaneously possible calls	Total traffic based on Erlang B model, Erl	Lines per upstream receiver	Total number of lines per upstream receiver	Lines per CMTS	Total number of lines per CMTS
G.711	10	115,2	40	29,01	414	591	1656	2364
	20	89,6	52	39,70	567	810	2268	3240
	30	85,3	55	42,41	605	864	2420	3456
G.728	10	57,6	81	66,29	947	1352	3788	5408

D. Example of VoIP service over HFC CATV network estimation

After the short description of the main parameters of voice telephony service, a scenario for VoIP service estimation will be considered. An example for such estimation is to evaluate the number of subscribers that are possible to be served at the same time. An area that is served by a CMTS consists of one downstream RF transmitter and four upstream RF receivers (such as Motorola BSR 1000) is assumed. The estimations are made for the following assumptions:

– Upstream modulation format: QPSK;
– Upstream channel bandwidth: *3,2MHz* (nominal data rate *4,6Mbps*);
– G.711 codec (Tables II and III);
– G.728 codec (Table 2);
– Blocking probability 1%;
– Traffic per residential user *0,07Erl* (*4,2min* call in average) ;
– *70%* of the CATV users are subscribed to VoIP service.

The estimations based on above assumed parameters are shown in Table VI.

IV. CONCLUSION

The numbers in Table VI are comparable to the traditional switched telephone networks. If the additional PHS options are applied, and low rate codecs or a 16-QAM modulation is used for the upstream channel (*3,2MHz@10,21Mbps*), than the VoIP network capacity will be greater than the switched telephone network capacity.

Lower rate encoding would realize a larger rate reduction for increased packetization periods. For example G.728 realizes nearly a *50%* flow rate reduction between *10ms* and *30ms* packetization compared to around *25%* for G.711.

IP-based telephony has a substantial disadvantage. Like all the IP-based services the telephony is very sensitive to packet loss and delays when the transmission rate is too low.

It is recommended to coordinate the choice of node and DOCSIS parameters so as to reach a target maximum PER for VoIP service while minimizing aggregate node and CMTS costs (as driven by utilization considerations). The maximum VoIP PER target should be chosen from a total VoIP packet loss rate target that also ac-counts for jitter-induced packet drops. The total VoIP packet loss rate target should be chosen according to voice service needs.

PER may increase significantly with packetization period if error conditions tend to be random in nature, such as from thermal noise or short impulse noise.

If or when VoIP and data traffic loading grows beyond the capacity of a current CMTS configuration, CMTS reconfiguration will be required. In this case optimal selection of maximum VoIP bandwidth allocation and the ratio of node-to-CMTS receiver assignment should be recomputed using new traffic load assumptions and parameter choices.

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