THE IMPROVEMENT OF END TO END DELAYS IN NETWORK MANAGEMENT SYSTEMUSING NETWORKCODING

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ABSTRACT

In this paper, we consider the application of network coding(NC) for network management system of Radio and Television Broadcasting Stations in wireless network using a narrow band radios modem &Routers) as a means of transmission to communicate between the loins broadcast TV/FM stations . Our main contribution is the application of NCimproved by the proposed an effective strategy,called Fast Forwarding Strategy (FFS) compared to a classical routing strategy with a level of providing guarantees of service quality (QoS) expressed in terms of reducing the End-to-End Delays (EED) from source to destination.The practical and theoretical study, carried out by the authors, show that the EED of proposed strategy outperforms that of the classical strategy.

KEYWORDS

Network Coding, Narrowband RF Networks, Delay, Encoding Strategy, Node Coding, Routing, Network computing

1. INTRODUCTION

Network delay is a performance characteristic of a computer network, telecommunications network or a network management system,which is essential to provide integrated broadcast solutions. The delay of a network specifies how long it takes for a bit of data to travel across the network from one node or endpoint to another. It is typically measured in multiples or fractions of seconds.

Network coding is a recently introduced paradigm for data dissemination in wireless networks able to increase throughput, reduce end-to-end delays, and enhance robustness

The benefits of network coding have been presented in various contexts. The authors of [1] have shown that a gain in speed and bandwidth can be obtained by using the coding system instead of traditional routing. In [2], two evaluations of the benefits of network coding are shown which can help to save bandwidth through the coding of information [3].
Several advantages of network coding are illustrated in an example given in [3] where one can see that the multicast transmission rate in the case of network coding is considerably larger than the transmission rate of multicast case of traditional routing.

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The coding system is not only used to save bandwidth and increase throughput[4]but it can also be useful for the robustness of the networkand performed End to End Delays , especially when the links in the networks fail, such as in wireless networks.

Another advantage of Network Coding in wireless networks is the possibility of reducing the amount of energy per bit, or in other words, the possibility of reducing the use of network resources compared to traditional routing solutions [5] [6] [7].

In addition, the advantage of coding network to access and store large files in peer-to-peer has been presented in [8]. It is shown that network coding can obtain a gain of about 10 times (with the use of codes) that with the transmission of information not encoded.

Therefore the codification of network can ameliorate considerably the bit rat, robustness, complicacy and the security of a network. [9][10][11].

In contrast to the store and forward paradigm, network coding implements a more complex store, encode, and forward approach where each node stores the incoming packets in its own buffer, and successively sends a combination of the stored data.

In view of the above explanation, we can see that network coding has various parameters: the manner to combine packets, the size of the base of the vector space of coefficients, the number of packets to be combined, etc...

However, reducing the number of packets to combine decreases the gains of network coding in terms of robustness and throughput, increase engenders a long delay in the application layer.
The maximum delay generated this strategy was evaluated at a node coding using the network calculus [12] [13].

The different guarantees and constraints characterizing the network and the flows can be represented by using the network calculus framework [13] which allows the computation of upper bounds in terms of delays, throughput or buffer sizing.

The narrow band and the wide band microwaves amplifiers are very used in the communication and detection systems (spatial telecommunication, radio communication, radar detection, control system …) [14].

We will measure End to End delay versus throughput for incoming flow / total network capacity, in Narrowband RF wireless network for management of TV and FM broadcast stations from source to destination, using Radio modem &Router unit as a means of wireless communication [15].

On the other hand, End-to-end delay is a key metric in evaluating the performance of networks as well as the quality of service perceived by end users.

So, we will apply an encoding strategy presented in [16] called Fast Forwarding Strategy for implementation of the End to End Delay. We use the concept of code block. Its main feature is that packets are allowed to leave the encoding nodes even if all this block via this node packages have not yet reached this node. This approach can reduce unnecessary wait times of packets in the routers. Finally, by way of comparison, the classical strategy to routing / multiplexing is also discussed in this paper.

This paper is arranged into seven sections including introduction. Section 2 gives a main of objectives of our experiment. Section 3 gives overview of Network calculus theory. Section 4 gives overview of the end to end delay .Section 5 describes the communication Network and the two strategies applied in this paper. Section 6 describes the experimental setup used to prove this relationship and real time hardware simulation results are presented. Conclusion and future works are presented in Section 7.

2. OBJECTIVE

The broadcasting TV/FM stations in rural mountainous are often located in sites of high altitude and the access is very difficult and sometimes inaccessible (during the bad weather) which has a high error rate.

In the case of a failure we do not know at what level to find the failure, in order to prepare the mission, in addition, these stations are isolated; their operation is not monitored nor operates or has remote control.

The goal of our application, in this paper , is to decrease worst case end to end delays (between source and destination) to better communicate with the broadcasting TV / FM station in order to conduct of the interrogation, operation, monitoring and remote management using Simple Network Management Protocol (SNMP) and a type of propagation line of Line of sight (LoS),and this will be accomplished with a new mode of data transfer in which the node of network can accomplish operations of codification on the data of a packet (Network coding).

The improvement of end to end delays (One of the advantages of network coding) between the source node (station) and the node destination (see figure1) be practiced in wirelessnetwork [15] using the characteristics of propagation network Narrow band radio modem & Router transceivers using random network coding, to manager devices, equipment, TV / FM stations i.e.: transmitter TV / FM, satellite receiver multiplexers, inverters, energy parameters etc..(See figure 2)

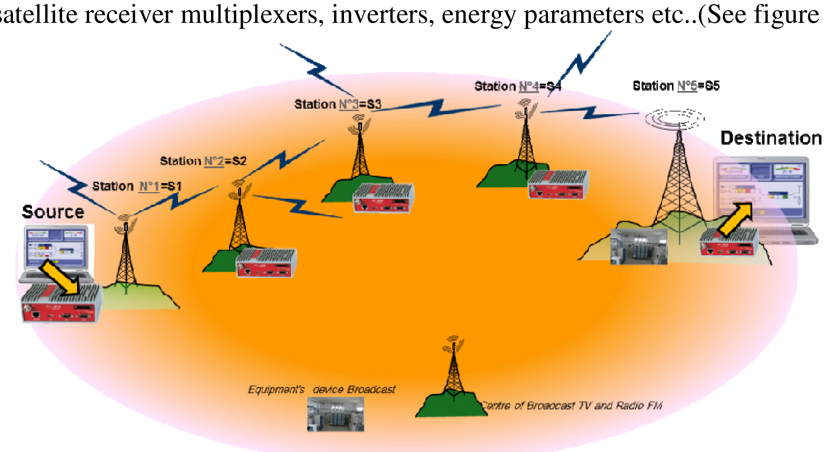

Figure 1.Network with multiple levels of coding / multiplexing

Thecontrol hardware (Server, client and practical application execution)at the station S5 is shown in Figure 2(Left).The real interface of application (of power studio SCADA Software, CIRCUTOR) is shown in figure 2(Right) ,from this interface we can control all the equipment station S5, namely diffusion (transmitter DVB-T/FM ..), transmission (SDH / PDH ..), energy (UPS , MT/Transformer) well as locals(Fig 2 Right).

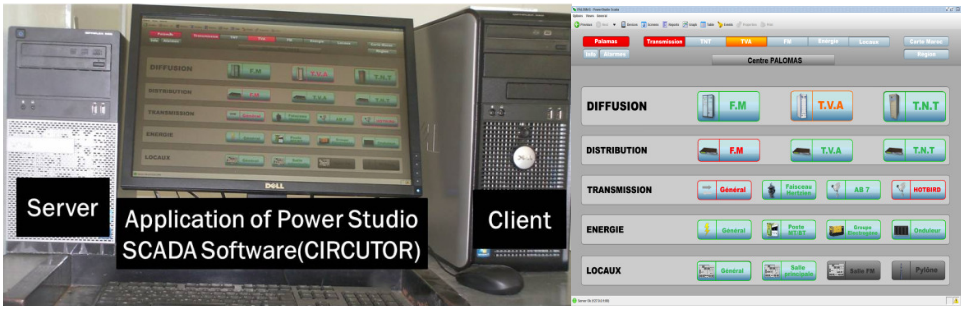

Figure 2 .(Left) : Control Hardware of broadcasting TV/FM Station S5 (Destinationstation "Palomas") see figure 1 and 6. (Right): Real Interface

3. NETWORK CALCULUS THEORY.

3.1. Notation

We first introduce the notations shown in Table 1.

Table 1: Notations

Parameters	**Notation**
R(t)	cumulative function
α	Stochastic arrival curve
β	Stochastic service curve
F	Data stream
$\otimes$	min-plus convolution
σ	regulation curve
R^{out}	output flow
$\gamma r,b(t)$	affine functions $\gamma r,b(t)$

3.2. Network Calculus (NC)

Network Calculus (NC) can be defined as a set of rules and results that can be used to compute bounds of performance parameters of communication networks. The most common parameters of interest are: end-to-end delay; maximum/minimum transmission rates and buffer usage.

NC is based on the idea that a detailed analysis of traffic flows is not required in order to specify a network performance, if the following conditions are satisfied:

- Input flows have limited burstiness.
- Some service guarantee is provided.

The above conditions define a minimum system for the NC (Figure 3):

- A filter to limit (or shape) the input traffic;
- A network that can offer some service guarantee.

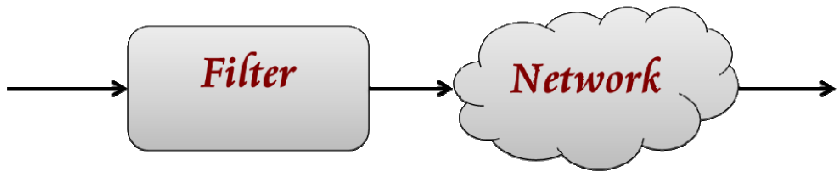

Figure3.A minimum system for the NC.

Network calculus is a theory that studies the relations between rows of data in a network. The movements of data are described by cumulative functions R(t) which counts the amount of data that arrives/departs from a network element up to time t.

A second type of functions that is used in network calculus is the arrival and service curves. These functions give some constraints to shape cumulative arrival in a server or to guaranty a minimal service of a server. These functions are used for computing worst-case delay bound.
In fact, Network Calculus is a framework providing deterministic bounds to end-to-end delays, backlogs and other QoS parameters by using the Min-Plus algebra. This theory was introduced and developed by Le Boudec and Thiran in [17] by generalizing previous works such as [18][19]. The following definitions and results are extracted from [17] where a detailed presentation of this theory can be found.

1) A data stream F transmitted on a link can be described by the *cumulative function* $R(t)$, such that for any $y > x$, $R(y) - R(x)$ is the quantity of data transmitted on this link during the time interval $[x, y]$.
2) Let F be a data stream with cumulative function $R(t)$. We say that an increasing function $\alpha$is an arrival curve of F (or equivalently f) if for any $0 \leq t_1 \leq t_2$, $R(t_2)-R(t_1) \leq \alpha(t_2 - t_1)$. A common class of arrival curves are the affine functions $\gamma_{r,b}(t) = rt + b$ for $t > 0$ and 0 otherwise. The curve $\gamma_{r,b}(t)$ represents the arrival curve of the leaky bucket controller with leak rate r and bucket size b.
3) The min-plus convolution of two functions X and Y is defined as $X(t) \otimes Y(t) = inf_{0 \leq s \leq t}(X(s) + Y(t - s))$It can be shown that $\alpha$is an arrival curve of R if and only if $R \leq R \otimes \alpha$.
4) Let R^{out}be the output flow of a node with one input flow R. We say that the node offers a service curve $\beta(t)$ to R if for any $t > 0$, $R^{out}(t) \geq R(t) \otimes \beta(t)$.
5) Assume that a flow $R(t)$, constrained by an arrival curve $\alpha(t)$ traverses a system offering a service curve of β. The output flow $R^{out}$is constrained by the arrival curve $\alpha \oslash \beta$, where $\alpha \oslash \beta = \sup_{v \geq 0}\{\alpha(t+v) - \beta(v)\}$.
6) The Burst Delay Service curve $\delta_T$is equal to ∞ if $t > T$ and 0 else.
7) The rate latency service curve $\beta_{R,T} = R[t - T]^+$ is equal to $R(t - T)$ if $t > T$ and 0 else.
8) The backlog of a flow R in the node at the time t is the amount of data «in transit" in the node. This backlog, defined as $R(t) - R^{out}(t)$ for all t, satisfies . $R(t) - R^{out}(t) \leq \sup > 0\{\alpha(s) - \beta(s)\}$

4. END -TO -END DELAY

It is easy to see that when a flow passes through coding nodes, it may become coupled with other flows after coding. To avoid the problem caused by flow coupling, a straightforward end-to-end delay analysis is to use the node-by-node analysis approach [20].

Nevertheless, node-by-node analysis will result in a loose bound [20, 21] scaling in O(n2 log n), where n is the number of nodes along an end-to-end path. In contrast, the end-to-end delay bound derived from the concatenation property is much tighter and scales in O(n log n) [20,21]. It is clear that we can-not directly use the property of node concatenation [21] to calculate the service of servers in tandem, because of the flow coupling along the path. Can we avoid the coupling problem without the sacrifice of the scaling property of end-to-end delay bound?

The answer is positive due to the following theorem. We use Si to denote a virtual server and $A_i^j, j=1,....,m_i$, to denote the input flows to the server, where m_i denotes the number of input links to S_i.

Theorem 1 (Ultimate Output Characterization) Consider an input flow A_1^1 passing through an end-to-end path, which consists of n virtual servers, $S_i, i=1,....,n$, in tandem. Assume that the output flow of A_1^1, $A_i^* = A_{i+1}^1, i=1,....,n-1$. For a virtual server S_i, if the number of its input links, m_i, is larger than 1, there may exist other input flows $A_i^j, j=2,....,m_i$, that are coded together with the flow A_i^1, according to a network code. Assume that $A_i^j, j=1,....,m_i$ has a stochastic arrival curve $A_i^j \sim mb < f_i^j \alpha_i^j >$. Assume that S_i provides to the input flows a stochastic service (including coding and transmission) curve $S_i \sim sc < g_i, \beta_i >$. The ultimate output flow $A_n^*(t)$ A_ n has an m.b.c. stochastic arrival curve $A_n^*(t) \approx_{mb} \langle f, \alpha \rangle$, where:

$$f(x) = ((...((\sum_{j=1}^{m_1} f_1^j) \otimes g_1) + \sum_{j=2}^{m_2} f_2^j) \otimes g_2 + ... + \sum_{j=2}^{m_n} f_n^j) \otimes g_n$$

$$\alpha = (...((((\alpha_1^1 V ... V \alpha_1^{m_1}) \otimes \beta_1) V ... V \alpha_2^{m_2}) \otimes \beta_2) V ... V \alpha_n^{m_n}) \otimes \beta_n$$

Proof: We can use the following algorithm to calculate the arrival curve of the ultimate output. Starting from the first virtual server, S1, we perform the following calculation:

- Step 1: We calculate the arrival curve of the output flow from current virtual server.
- Step 2: Move to the next virtual server along the path.
- Step 3: Repeat Step 1 until the output from the last virtual server is calculatedaftersimple manipulation with the above algorithm,

Theorem 2 (End-to-End Delay) Consider an input flow $A_1^1 \sim mb < f_1^1, \alpha_1^1 >$ passing through an end-to-end path, which has the same settings as in Theorem 1. Assume that the ultimate output flow $A^* \sim mb < f, \alpha >$, where f and α can be obtained with Theorem 1.

Also assume that at the destination, the decoding service recovering the traffic belonging to A_1^1 follows $\sim_{sc} < \phi_1(f), \phi_2(\alpha) >$, where ϕ_1 and ϕ_2 are functions of f and α, respectively. The end-to-end delay of A_1^1 at time t satisfies: for all t ≥ 0 and all x ≥ 0,

$\Pr ob\{D(t) > h(\alpha_1^1 + x, \phi_2(\alpha))\} \leq f_1^1 \otimes \phi_1(g)(x)$, where $h(\alpha, \beta)$ is the maximum horizontal distance between functions α and β, which is defined as $h(\alpha,\beta) = \sup_{s \geq 0}\{\inf\{\tau \geq 0 : \alpha(s) \leq \beta(s+\tau)\}\}$ **Remark**:

Although our calculation is node-by-node, the end-to-end delay bound is much tighter than that obtained by the node-by-node analysis described in [7, 12]. This is because the node-by-node analysis derives the delays at each individual server which are summed up as the end-to-end delay. In contrast, our calculation has the same flavor as in the end-to-end delay analysis based on the concatenation property [7, 12], which only considers the input and the ultimate output from the system.

5. NETWORK

5.1. Communication Network

Consider a communication network represented by an acyclic directed graph G = (V, E), where *V* is the set of network nodes and E is the set of directed links , between network nodes ,with a vertex set $V = \{v_1,.....,v_m\}$ and an edge set E. The directed edge connecting the node v_i to the node v_j is denoted by $e_{i,j}$.

We assume that all the nodes are synchronized. Each edge $e_{i,j}$ has a capacity $C_{i,j}$ (bits/sec), meaning that a packet of L bits is transmitted in at least $L/C_{i,j}$ seconds. Since the system is assumed to provide QoS guarantees, we consider that, for each edge $e_{i,j}$, the maximum transmission delay of a packet of L bits is known and equal to $L/C_{i,j} + T_{i,j} = w_{i,j} + T_{i,j}$. In other words, the edge $e_{i,j}$ has the rate latency service curve $\beta_{C_{i,j}}, T_{i,j}(t)$. We suppose that the capacity of every output edge of a node is greater or equal than the sum of capacities of all input edges. This hypothesis is used to be fair with the routing approach, but for network coding, it is sufficient to have the output capacity greater than the maximum of the input capacities.

We define an oriented link between nodes v_i to node v_j by $e_{i,j}$. Each link $e_{i,j}$ has a capacity $C_{i,j}$ (bits / sec), which means that a packet of L bits will be transmitted in $L/C_{i,j}$ seconds, where L denotes the packet length.

As the system must provide QoS guarantees, we consider that, for each link $e_{i,j}$, the maximum of a packet transmission delay of L bits is known and equal to $L/C_{i,j} + T_{i,j} = w_{i,j} + T_{i,j}$ .In other words, the link $e_{i,j}$ a curve rate-latency service $\beta_{C_{i,j}}, T_{i,j}(t)$. We assume that the capacity of the outgoing link is greater than or equal to the maximum entry capacity. In other words, the ability of outgoing link must withstand, without congestion, all flows of input links. Classically, with network coding, just that outgoing link has a capacity of the order of the maximum throughput of the input stream (ie d. Worst case, the outgoing link capacity must be greater than or capacity equal to the maximum capacity of input links).

Flows generated by sources are composed of fixed-length packets L. They satisfy two constraints. The first is related to the notion of block. We assume that all sources cut the time interval $[t_i, t_i + \Delta]$ Δ fixed length. In each of these time intervals, each source generates at most one packet. All packets generated by different sources in the same time interval $[t_i, t_i + \Delta]$ are the set of information packets of the fifth block (code word). While some sources do not generate packets in this time interval, the packet of information from this source is considered invalid.

Level flow, it should be noted that the fact that the sources generate at most one packet per time interval imposes a constraint on the rate and degree of variability of the flow. Indeed, with this constraint, the maximum flow is $\rho = L / \Delta$.

5.2. Classical strategy

This strategy is based on the classical definition of the network coding. Let us consider an intermediate node with *n* input Flows and one output Flow (see Figure 4).

We consider that for each generation i, a deadline of the arrival time of P_i is known. , the linear combination corresponding to a generation i is done as soon as, for all the input flows, at least one of the following points is verified:

- All the packets of the generation are in the buffers.

- Some packets of the generation i are not in the buffers and the deadline of the arrival time of the generation i is exceeded or their corresponding packet of the generation i +1is in the buffer.

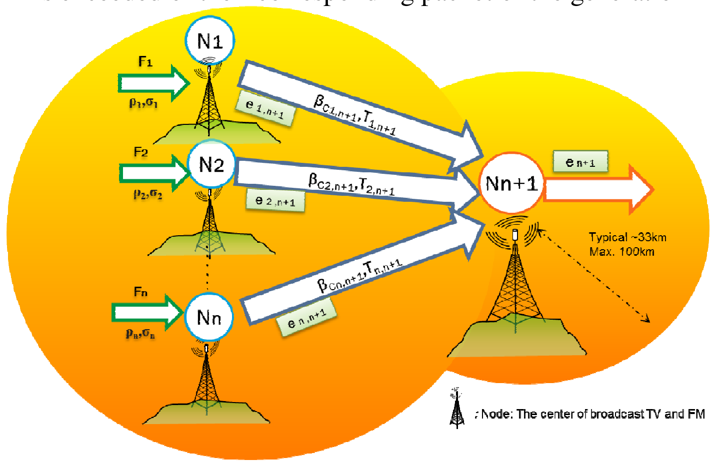

Figure 4: Node with n input stream

The last point indicates that the corresponding flow does not contain a packet of the generation i. In this case, the linear combination is only done with the packets of the generation i present in the node. Algebraically, this is equivalent to replacing the missing packets by packets full of zeros.

5.3. Fast forwarding strategy at the intermediate nodes

The system was designed to work with a given number of flows and is optimal when all the flows are active.

When some of the flows are idle, the others flows wait them in the coding nodes and consequently, their end-to-end delays are increased. The improvements we propose allow avoiding this problem by authorizing the packets to leave the coding node even if the whole generation is not arrived. This strategy is *called fast forwarding*.

Let us consider an intermediate node with n input flows and one output flow (see Figure 5) with the network hypotheses precedents.

Suppose that a packet of a given generation X arrives at the coding node $n + 1$ at time t. The fast forwarding strategy of this coding node is the following:

If the buffer is empty, the packet is multiplied by the finite field coefficient determined by the network code and is transmitted over the output link (if this link is not used by another packet transmission started before time t).

If the buffer is not empty:

- If there is not a packet of the generation X in the buffer, the packet is multiplied by its corresponding finite field coefficient and added at the end of the buffer. For example, on Figure 5, the packet P_3^1 arriving from node $N1$ is added at the end of the buffer.
- If there is a packet of the generation X in the buffer, the arriving packet is multiplied by its corresponding finite field coefficient and is directly summed to the packet of the generation in the buffer. For example, on Figure 5, the packet P_5^2 arriving from node N_2 is summed to the packet P_5^1 already present in the buffer.

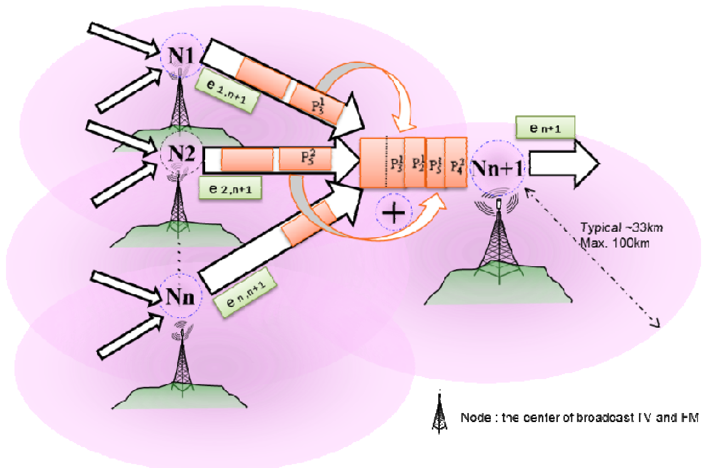

Figure 5. Fast forwarding strategy

Note that, this strategy could lead to generation desequencing (like in the Figure 5).

To estimate the end-to-end delays and the buffer size, we must determine the maximum delay suffered by a packet in an intermediate node. From the strategy described previously, it can be deduced that a packet must wait at most the time needed to transmit the maximal number of different generations which can be found simultaneously in the intermediate node (when the packet arrives at the node). The arrival time at each intermediate node and the intergeneration times are used to calculate this number

6. APPLICATION

6.1. Case study of end to-end delay bounds

In this section we apply the network calculus formulation to the derivation of end-to-end delay bounds.

The main goal is to illustrate the derivation of the bounds in two different scenarios depending on two different types of statistical independence assumptions.

In our case study, we propose a network of real application with multiple levels of coding/ multiplexing. Concretely, we consider the tandem network with cross traffic from Figure 6. A through flow traverses five nodes and each node is also transited by a cross flow; the notation for the flows is as in the figure 6. Each node has capacity C and serves the packets in a static-priority (SP) manner giving the cross flow's packets higher priorities.

Let us consider the network presented in Figure 6. In this case, sources 1,2 and source 3 multicast the flows F1, F2 and F3 towards two receivers R1 and R2 which are the center of broadcasting TV / FM. This network contains three levels of coding / multiplexing. Each level of coding / multiplexing has an impact on the time maximum end-to-end. The results illustrate the advantage of coding strategies in networks that contain multiple streams flow and several important levels of packet processing.

The following figure 6 shows a real part us and practice in the region in which apply us the work of his paper. We will present the results directly on the maximum period of end-to-end. The results can be calculated as follows:

- F1, F2 and F3 are constrained by the same affine arrival curve $\alpha_{\rho,\sigma}(t)$.

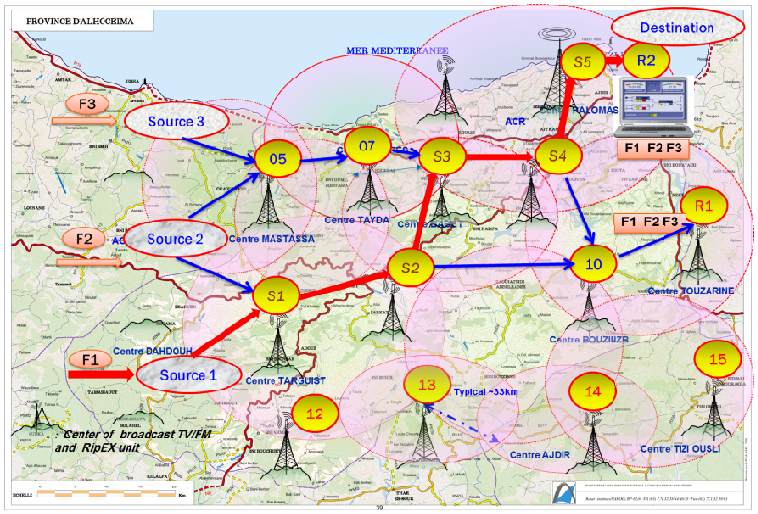

Figure 6 - Network real with multiple levels of coding / multiplexing.

- All links have capacity C except the input links of the receivers which have capacity C_{out}. It must be noted that under these hypotheses, network coding does not improve the throughput compared to the multiplexing approach. We have σ = 2 packets and $\rho = \frac{L}{C} \leq C$.
- All links $e_{i,j}$ have also the same service delay $T_{i,j}$=T. Therefore each link $e_{i,j}$ provides a service delay of a packet of L bits is known and equal to L /C+ T = ω + T.
- We also assume that each node in the routing / multiplexing N_k offers a service curve $\beta_{C,\tau_k}(t)$ or $\beta_{C_{out},\tau_k}(t)$ where τ_K is the delay of service offered to the total flow.
- Similarly, each intermediate node N_k offers a service curve $\beta_{C,T_B^k+T_{lc}+\tau_k}(t) = C\,(t + T_B^K - \tau_k)$ or $\beta_{C_{out},T_B^k+T_{lc}+\tau_k}(t) = C_{out}(t + T_B^K - \tau_k)$

Where T_B^K is the maximum time, spent by a packet in the buffers while waiting for corresponding packets of others flows , T_{lc} denotes the maximum time needed to achieve a linear combination of packets and τ_K is the service delay to transmit a packet.

With the conditions described previously, the worst case delay for multiplexing and coding cases is obtained on the paths crossing the maximum of nodes, i.e. for paths crossing five nodes witch are the same property, we can choose one of them and study its worst case delay. We choose the path from Source 1 to R2 which cross nodes (stations) S1, S2.S3, S4 (Rural areas) and S5 (Urban area).

Each source transmits in multicast packets to all receivers. The sources share the same clock, but they do not produce their packets simultaneously. The length of a block, Δ, varies from 10ms to 50ms. Thus, the flow rates vary from 20 to 100pps (packets per second).

All links in the network have the same capacitance C, which is equal to 200pps and the delay experienced by a packet on a link $e_{i,j}$ is comprised between L/C and $L/C + T_{i,j}$.

The Value $T_{i\,j}$, corresponding to the transmission time of a packet on a link is randomly following a uniform distribution in the interval [0, 10] ms**.**

All nodes in the network have the same service time τ which follows a uniform distribution in the interval [0, 15]. This period is on average equal to 7.5 ms.

T_{lc} refers to the maximum time required to achieve to achieve a linear combination of packets, is considered very small and negligible with respect to other.

The numerical values are taken as following:

Table 2: Notation and value

Parameter	Value
T	10ms
τ	15ms
Δ	10ms
w	5
C	200pps
Cout	200pps
T_{lc}	0
L	1000 bit

6.2.EED Measurement in different values of Throughput (pps) of incoming flow Network.

The aim of this experiment is to measure the EED of the Radio modem &router as a function of the incoming flow to gain a sense as how does the incoming flow affect the EED.

In our practical application, the topology showed in Fig. 6 is well-respected. Radio modem &Router units module configured as a source S1 sends packets to destination R2.

Various EED measurements areeffectuated, in comparison to different values of the throughput (pps) of incoming flow Network. Therefore, to avoid reception overcharge with the SCADA communication protocol, we used a lower value of throughput of incoming flow Network up to 65 pps, in this case the shape of measurements EED curve is shown as a function of throughput of incoming flow.

We will broach two strategies measurements:Classical strategy (CS)and Fast Forwarding Strategy (FFS).

6.2.1. Theoretical study of (CS)

To determine the maximum worst case response from end to end delay using the techniques of Network Calculus [18].With the conditions described previously, the worst case delay for multiplexing and coding cases is obtained on the paths crossing the maximum of nodes, i.e. for paths crossing 5 nodes.

We choose the path from *Source* 1 to *R2* which cross nodes (stations) S1*, S2, S3, S4* and S5.
In the multiplexing strategy (classical strategy), the maximum delay of flow F1 at the output of multiplexer of the station S1 can be obtained by simplyapplying the formula given in previous sections. In the general, classical, coding strategy, with the results of the maximum delays given by equation in [16] (versus of the incoming flows in the network) of the (CS) ,calculated on

the path from Source 1 to R_2 that passes through nodes S1,S2,S3,S4 and S5 (path marked in red in Figure 6) ,are shown in Figures 7.

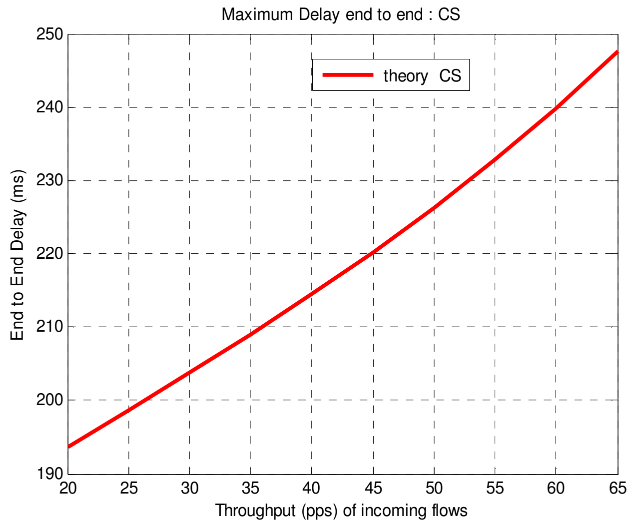

Figure 7: Maximum Delay end to End (of CS) to end versus throughput of incoming flows of the network.

We see that end-to-end delay increases significantly versus of incoming flows of our network scheme.

6.2.2. Practical study of (CS)

6.2.2.1. First method practice more calculating.

Before all was discussed in advance the preliminary study to describe each site TV/FM in the region includes:

- Geographical coordinates.
- Possibility of visibility between sites.
- Broadcast and reception frequencies.

From the parameters aforementioned and through the software package CHIRplus_BC the useful information can be draw such as the distances and shows the possibility of direct visits (in LOS from the sites) between the broadcasting TV/FM stations telemetered see Figure 8.

CHIRplus can be used by operators as well as by regulators to analyze existing networks, plan new frequencies, or perform the necessary coordination calculations according to international agreements.Maximum Distance is calculated by CHIRplus show Figure 8

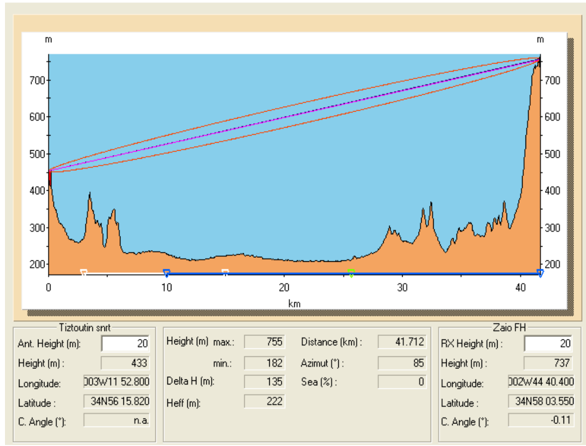

Figure 8. Maximum Distance in LOS between two stations in our case study

Consider there is N=5 nodes (stations) between the Source1 and the Destination see figure 6.Therefore, each packet transmitted between source (S_1) and destination (R_2) must traverse more (five) communication links in order to reach the final destination.

The end-to -end delay is actually derived from the nodal delay, i.e., the delay at the single router. The end-to-end delay for N nodes between the Source Host and the Destination Host is as follows, $D_{end-end} = N(D_{proc} + D_{trans} + D_{prop} + D_{queue})$ (1)

Where,

Table 3. Parameters and notation

Parameters	Notation
$D_{end-end}$	End-to-End delay
N	N is the number of nodes between the sender and the receiver.
D_{proc}	Processing delay at each Router
D_{trans}	Transmission delay
D_{prop}	Propagation Delay
D_{queue}	The Average Queue delay

Let the value of $D_{end-end}$ denotes packet-delay (we sometime refer it as link delay) that is associated with each direct communication link. Therefore, each transmitted packet will typically experience a delay of $D_{end-end}$ on a particular link. The delay includes transmission, processing, and average queue [22] and propagation delays such as (1).

In connection less communication such as IP network, there might be multiple routes exist between a pair of source and destination. As a result, each packet might follow a different route in order to reach the final destination where each route requires traversing of one or more communication links (6 links) see figure 6. A single route between a pair of source and destination can be defined as: link e_{S_1,R_2} .

The transmission delay between source S1 and destination R2 is,

$$D_{Trans} = \sum_{i=0}^{N} D_{tran,i} = \sum \frac{L_i}{R_i}$$

Where,

Ri= Rate is transmission rate of router i (bits/sec).

Li= Packet Size of router i.

The propagation delay between two routers generally ranges from $2*10^8$ m/sec to $3*10^8$ m/sec.

That is the Propagation Delay,

$$D_{\Pr op} = \sum_{i=0}^{N} D_{prop,i} = \sum_{i=0}^{N} \frac{d_i}{s_i}$$

Where,

d= distance between two routers

s= propagation speed of the link

We note that D_{proc} and D_{queue} [22] are taken in our application as delays in the worst case.

The measurements and calculations have resulted to (Figure 9):

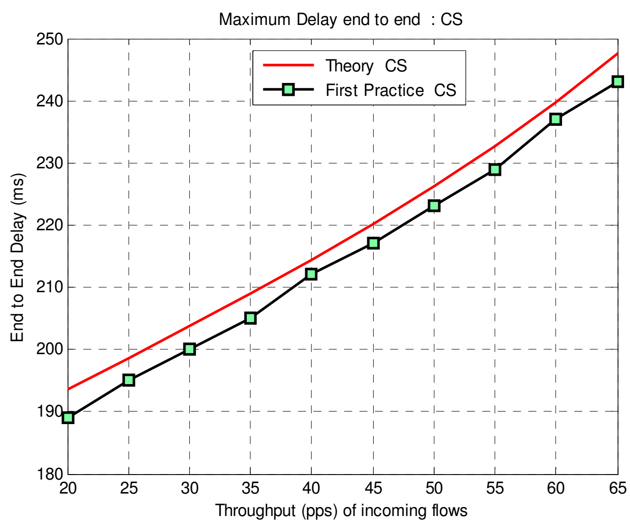

Figure 9. Maximum Delay end to end: Comparison between theory of (CS) and measurement Practice more calculation of (CS).

We conclude that the both measurement results (first method) and theory results are almost the same.

6.2.2.2. Second method practice

The hardware system including computer software used is:

- PowerStudio SCADA Software: PowerStudio SCADA, in conjunction with CIRCUTOR equipment and systems, adapts to particular needs by providing tools for the supervision and control of the installations of the equipment's of broadcasting.
- Five radio modem &routers function Works in Narrowband (out power 10 watts) which are characterized by the SNMP management that will support the base MIB (Management Information Base) SNMP (Simple Network Management Protocol) Protocol through the MIB browser.

- Radio modem &Router use the same band VHF /UHF (band IV and V), VHF/UHF bands 350 MHz this band is somewhere between 160 and 450 MHz. reserved to broadcasters have highly favorable propagation characteristics. Penetrating through foliage and structures, they reach far and wide distance more than Wimax [23, 24].
- We can use the omnidirectional antenna KA160.3 which is designed for base radio stations working in bands of 158-174 MHz The antenna ,used in our application, has an Omni-directional radiation pattern with the gain of 3 dB and is adapted for the top-mounting. The antenna is broadband and that is why it is well-chosen for duplex operations.
- We can include in each Router unit CAP (Chip Authentication Protocol) for More Secure Authentication.
- R&S®FSQ Signal Analyzer (Figure 10) : that is capable of supported technology application applied in this paper (to perform the necessary measurements of the signals, receipts and issued, of each node).

Figure 10.R&S®FSQ Signal Analyzer (Left) and RipEX Radio modem &Router.

- We can use the same antenna system that is already used by the broadcast DTT system.
- We will use the free channels abandoned by analog television.
- The values received at the level of each site vary between 38 and 70 dBµV what is
- recommended to plug user for correct operation of household appliances for the bands IV and V.

The units Source1, S1, S2, S3, S4 and S5 are all of the same type Routers are all identical with regards to hardware and software configuration.

We are in the condition is a condition where a signal travels over the air directly from a wireless transmitter to a wireless receiver without passing an obstruction Line-of-sight (LOS), because in LOS environment, signal can reach longer distance with better signal strength and higher throughput.

It can also measure the EED; packets are sent from the source to the destination using the ping utility, over different route lengths and beaconing intervals. The EED is taken as one-half the RTT(Round trip time) using **the same**RTT [25] path round-trip (see Figure 1and 6) to minimize the difference between a request for data and the complete return or display of that data.

Thus the measurement results (second method) compared with last results are given in fig 12:

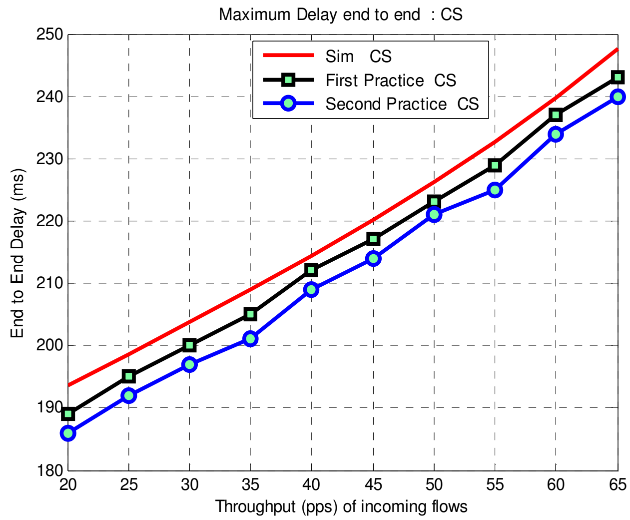

Figure 11: 'Maximum Delay end to end: Network with multiple levels of encoding / multiplexing.

We conclude that the both measurement results (first, second method) and theory results of CS are almost the same.

6.2.3 FFS: Fast Forwarding Strategy

The maximum delays of FFS given by equation in [16] ,calculated on the path from S1 to R2 that passes through nodes S1,S2,S3,S4 and S51 (path marked in red in Figure 6) ,are shown in Figures 12.

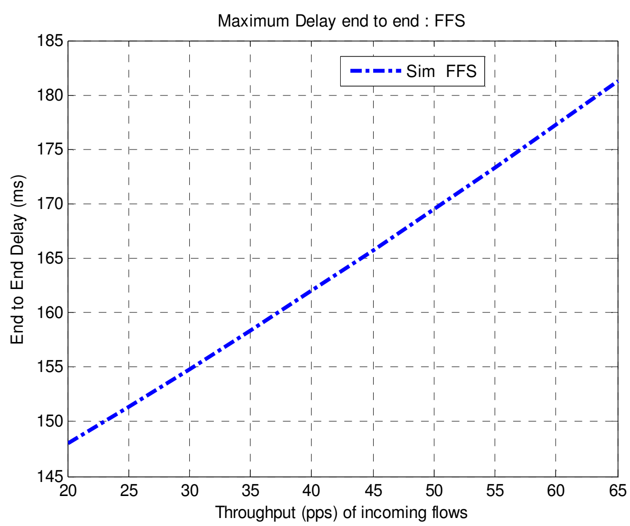

Figure 12: Maximum Delay end to end of FFS: Network with multiple levels of coding / multiplexing.

The theoretical and the practical comparison between (CS) and (FFS) show us:

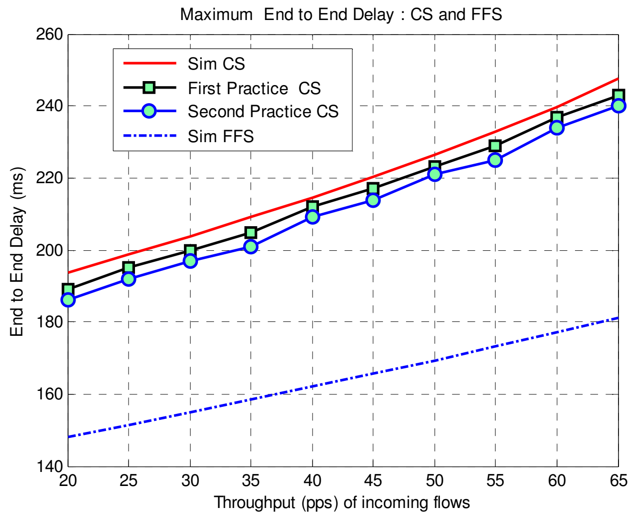

Figure 13:Maximum Delay end to end: Network with multiple levels of encoding/ multiplexing

6.3. EED Measurement in different values of total Network capacity.

We measured the EED performance of the Routers in output network;we compare the results EEDmeasurements of two strategies CS and FFS versus total network capacity. Network topology in Fig. 6 is considered where the packets transmitted from the Source 1 to R2 using five (Radio modem & Router).

Data throughput measurements were carried by means Ethernet interface ETH TCP/IP (between device and Router). The results of FFS theory is given by equation in [16].

The network coding with FFS shows an improvement delay, approximately 230(ms) until 300 (ms), versus total network capacity packet per second (pps) which varies between 20 and 65 (pps), see figure 14.

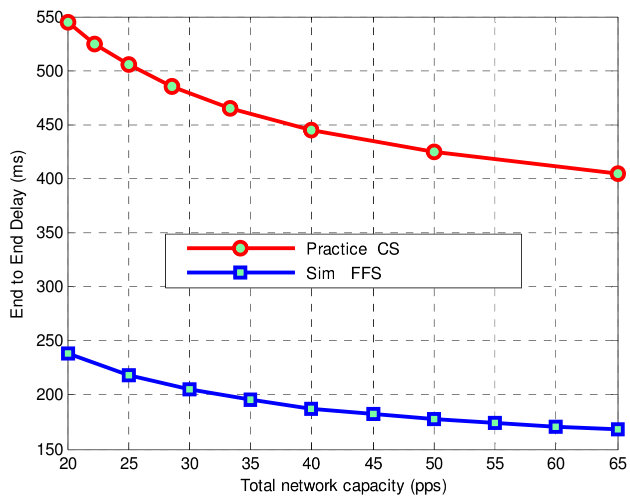

Figure 14: Maximum Delay end to end versus total network capacity (pps)): Network with multiple levels of encoding / multiplexing.

Network coding using FFS improves (decrease) well the end to end delays, when the total network capacity (pps) increases, see figure 14.

Note: It is interesting to note that The use of a discrete-event network simulator 'ns-3 'and of 'JiST' ,that runs over a standard Java virtual machine, given the almost same results as the practical and theoretical study.

6.4. Discussion

The comparison of the worst case delays of the two coding strategies directly shows that fast forwarding strategy obtains better end-to-end delay bounds than general coding strategy.
The end-to-end delay bound obtained with fast forwarding strategy is better than with multiplexing strategy if and only if:

The conclusion of this comparison depends on the relationships between the different parameters. Unsurprisingly, the performance of network coding strongly depends on the value of T_{lc}, which is the delay due to a combination of two packets.

For a fixed T_{lc}, the interest of network coding grows when the parameters τ and T are increased. These parameters are respectively the service delay of a node and the transmission delay on a link. Note the coefficient of T is strictly greater than 0 because the time needed to send a packet (L/C) is necessarily lower than Δ which is the duration of a generation range. It can also be observed (with the parameter σ) that the more the traffic is bursty, the more network coding is better.

The fast forwarding strategy gets the same gains in terms of average delay. This strategy can be used in all networks where the code is fixed.

7. CONCLUSIONS

This paper has presented two network coding strategies for networks providing QoS guarantees. These two strategies are evaluated in terms of maximum delays for a packet to be treated by a node. To reach the final results we have presented the relationship between different parameters (such as coding delay, transmission delay, throughput, burstiness, generation duration, . . .) in order to determine in which conditions the network coding allows to decrease end-to-end delays guarantees.

The theoretical and the practical comparison between End to End Delay ms of CS and FFS versus throughput of the incoming flows of packet (packet per second) (pps), show as Fig 13.
The figure 14 shows the theoretical and the practical comparison between End to End Delays (ms) of CS and FFS versus total network capacity.

The Comparisons of the maximum delay of two strategies, either at a node or at a network show that the fast forwarding strategy FFS usually offers Delay End-to-End better than those offered by the classical strategy CS.

The future work consists to introduce new techniques and news methods for broadcast network in order that the routers send the signal over the same frequency channel,single-frequency network or SFN.

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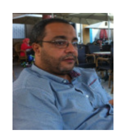

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