A New Distributed Application and Network Layer Protocol for VoIP in Mobile Ad Hoc Networks

Floriano De Rango, *Member*, IEEE, Peppino Fazio, *Student Member*, IEEE, Francesco Scarcello, Francesco Conte

Abstract – *In this work a new protocol for Voice over IP (VoIP) transmissions in wireless ad-hoc networks is proposed. Distributed architecture is necessary when dealing with dynamic environments, such as ports or battlefields, where creating infrastructures becomes expensive or impossible. Mobile Ad-hoc NETworks (MANETs) are based on a peer-to-peer approach and each node participates in the organization of the whole network. VoIP over MANETs is a challenging issue due to the intrinsic distributed nature of the existing peer-to-peer paradigm. This paper proposes a new protocol, capable of ensuring a Quality of Service (QoS) level for VoIP calls over a MANET and to manage a higher number of calls in the system. Novel metric and utility functions are proposed to perform the best path selection from source to destination nodes, respecting the QoS parameters for VoIP quality. In particular, an objective metric such as R-factor is considered and a flexibility index is defined, in order to maximize the number of acceptable VoIP calls. Performance evaluation shows that the proposed approach led to better network management in terms of admitted calls and respected QoS constraints.*

Index Terms **– VoIP, MANET, QoS, distributed architecture, peer-to-peer, IP-telephony, hostile environments.**

1. INTRODUCTION

Nowadays, an increasing number of people use wireless applications and VoIP [1], in order to make good quality and low cost calls. Typically, wireless technology is used only on the network segment that connects the end-user with the wireless interface, which forwards wireless packets to a wired backbone. Unfortunately, the employment of an infrastructure is often not possible in a distributed scenario, because classical QoS metrics may cause congestion or service disruption. In this paper an objective route selection metric based on the E-Model [2] is proposed, together with a suitable flexibility-based route ordering. The final goal is to overcome drawbacks typical of traditional approaches in routing strategies applied to distributed wireless systems, and to offer good call quality, even in dynamic and distributed networks.

An analysis of currently used VoIP systems (based on H.323 and SIP architectures) [3] shows how they are characterized by a set of fixed nodes (stateless/stateful proxies and registrar servers), which act as intermediaries between their endpoints or provide registration and localization of nodes. This kind of approach has some disadvantages that make it unsuitable when system dimensions grow:

- low fault-tolerance (if a proxy is damaged, the whole system will not work);
- low scalability in the number of supported parallel calls.

The situation degrades when hostile areas (where an infrastructure cannot be employed) are considered. Traditional VoIP structures become inadequate, because of the need for different fixed nodes [4]. If we consider a scenario with close endpoints and a far proxy then, without an infrastructure, the QoS levels become unacceptable: although a multi-hop protocol may forward the information to the proxy, the degradation introduced by each hop determines that constraints on the Mean Opinion Score (MOS) [5] are not respected, with consequent low-quality admitted calls. The proposed solution is based on a novel metric related to an objective measure of the QoS for calls [6] and on optimal codec selection in the route discovery phase of the wireless routing protocols for MANET, such

F.De Rango is with the Department of Electronic, Informatic and Systems (DEIS), University of Calabria, Arcavacata di Rende (CS), 87036, ITALY. E-mail: derango@deis.unical.it; homepage: http://culture.deis.unical.it/derango; phone-number: +390984494706.

P.Fazio is with the Department of Electronic, Informatic and Systems (DEIS), University of Calabria, Arcavacata di Rende (CS), 87036, ITALY. E-mail: pfazio@deis.unical.it; homepage: http://culture.deis.unical.it/fazio; phone-number: +390984494660.

F.Scarcello is with the Department of Electronic, Informatic and Systems (DEIS), University of Calabria, Arcavacata di Rende (CS), 87036, ITALY. E-mail: scarcello@deis.unical.it; phone: +390984494752.

F.Conte is with T.T. Technological Temptation SCARL, 00012, Guidonia Montecelio (Rome) , ITALY. E-mail: frankonte@gmail.com.

as Ad-hoc On-demand Distance Vector (AODV) and Optimized Link State Routing (OLSR) [7-9], where each node behaves actively and passively for routing operations. The new functionalities of the proposed system are:

- managing the construction of the lowest-cost path from source to destination in the best way on the basis of an objective measure of the QoS of the calls;
- dynamic codec selection strategy, in order to guarantee the best quality for new incoming calls, without degrading system performance;
- call admission control procedure integrated in the route selection, to refuse or direct on alternative paths the additional calls that can degrade the VoIP QoS constraints;
- route selection based on a suitable flexibility index, which allows the system to maximize the number of admissible new calls with the available resources, and hence to scale with the network size increase.

This paper is organized as follows: section II describes the state of the art on VoIP traffic management and metrics in MANET environment; section III defines the signaling protocol SIP, enhanced to fit our purpose in distributed wireless systems, the novel E-model based metric, and the optimization problem formulated for the optimal path selection; section IV focuses on the performance evaluation of the proposed solution considering two reference scenarios; finally, section V concludes the paper by summarizing the conclusions.

2. PRELIMINARIES AND RELATED WORK

The challenge in VoIP applications regards the possibility of offering customers a service that can be compared with the traditional telephone system [5]. In contrast to the Public Switched Telephone Network (PSTN), where an end-to-end connection is established when originating a call, packets networks use *statistical multiplexing* of network resources. Although sharing network resources among a multitude of users offers restricted and contained costs (that is the first prerogative of VoIP traffic), the use of common resources on the networks heavily affects the QoS perceived by users and some objective metrics for route selection and criteria for acceptance of calls need to be designed. Different parameters are used to describe the service offered by a system, such as delay, jitter, packet loss, echo and call setup delay.

2.1 Subjective and Objective QoS Evaluation

Subjective measurements of the QoS are carried out by a group of people (test subjects) [10-12]. A test phrase is recorded and, then, test subjects listen to it in different conditions. These tests are performed in special rooms, with background noise and other environmental factors, that are kept controlled for test executions. Some examples are: conversational opinion test, listening opinion test, interview and survey test. The subjects have to report their opinion on a scale chosen among those recommended by ITU-T, and the arithmetic mean of these results, called the *Mean Conversation-opinion Score* (MOS), is computed and evaluated. Once the absence of anomalies is verified, it is possible to continue with the next experiment.

Quantal-response detectability tests, defined by ITU-T Rec. P800 Annex C [12], represent the best method for obtaining information on analog sound properties (that are parameters that influence QoS) and they estimate voice quality through a vote on a scale called *Detectability opinion* scale, thus providing an opinion on the parameters that are tested.

The subjective methods are not practicable during the network planning phase because they are limited, impracticable and too expensive. In order to avoid these problems, new methods that permit the calculation of values representing the different damaging factor combinations of the network have been developed. The quality estimation, providing results as near as possible to MOS values, is the primary task of these methods. ITU proposes an objective, automatic and replicable testing method that takes into account the perceptual QoS [13, 14]. The objective measurement techniques use an approach where a voice sample represents the input signal to produce a score, representing the original signal produced by the network. Three objective evaluation methods can be distinguished:

- comparison methods: they compare a signal with a reference;
- absolute methods: they are based on the absolute quality estimation;
- transmission methods: they obtain a value through the network study and analysis in order to know the audio quality in advance [13, 14].

2.2 E-Model and PESQ

The most popular objective measurement method is E-Model, which belongs to the *non intrusive methods*, i.e. those measurement techniques that do not require an injection of additional traffic or probing to evaluate the quality of the original speech signal. E-Model was developed by an ETSI work group chosen by ITU [2, 15]. E-Model [15] is a computational method based on the assumption that each quality degradation type is associated with a certain type of damaging factor. It uses transmission parameters to predict the subjective speech quality of packetized voice. The primary output from the E-Model is the "Rating Factor" R [2, 15-18], and R can be further transformed to give estimates of customer opinion by mapping it to the MOS scale. The model calculates a base value for evaluating the quality determined by network factors. Each damage factor is then expressed in a value that is later subtracted from the base value. The main damage factors involved in the R factor computation are: Room noise (PS), background noise, noises deriving from microphone and loudspeaker use (Ds-Factor and Dr-Factor), Quantizing Distortion (qdu), Equipment impairment factor (Ie), Packet-loss probability (Ppl), Mean one-way delay (T), Absolute Delay (T), and Expectation factor (A). E-Model input consists of parameters that are valid in network design and installation. Some of these parameters are measured, while other parameters are extrapolated by standard reference samples. The first output of E-Model is Transmission rating factor R that is used for evaluating the quality perceived by the

network. The precise expression and details regarding the *R*-factor will be given in section III. Typically the range used for the *R*-factor is from 50 to 90.
The Perceptual Evaluation of Speech Quality (PESQ) is the last standard proposed by ITU (Rec.862, Feb.2001) [19] for the objective evaluation of the coded voice signal that goes through the telephone network. PESQ adds novel factors and methods to calculate signal distortion offering the possibility to use natural and artificial audio samples [19-22]. This model is recommended for the impact evaluation of a codec on quality, and for prototype network testing. It is applied for the evaluation of Transcoding, Transmission Errors on the transmission channel, Codec Errors, Noise introduced by the system, Packet loss, and Time clipping.

2.3 SIP Protocol over MANET

The Session Initiation Protocol (SIP) architecture [3, 23] is based on centralized entities. Two logical elements play a key role in the architecture, *registrar* and *proxy* servers. Registrars are the SIP entities where SIP users register their contact information once they connect to the SIP network. In a basic registration scenario, a SIP user agent communicates to its registrar server (the registrar IP address is usually preconfigured) the SIP user name of the user(s) using the device, referred to as SIP *address of records* (AOR) for that user, and the addresses where the user is reachable. Contact information is usually stored in the form of IP addresses or resolvable names, but other kinds of contact information, such as telephone numbers, can also be used. An association between a SIP AOR and a contact address is called a *binding*. SIP registrars exploit an abstract service, called *location service*, and return the bindings for the SIP AORs falling under their domain of competence to the SIP entities issuing a binding retrieval request. Proxy servers are needed because SIP users do not know, in the majority of cases, the current complete contact information of the caller but only the AOR. SIP presupposes that the AOR (SIP user ID) of the party to be contacted is known in advance, analogously to what happens when sending instant messages or e-mails. A basic SIP session involves the calling user agent contacting the calling side *proxy server*, which in turn will forward the message to the proxy server responsible for the domain of the called *user agent*. The called side *proxy server* retrieves the bindings for the called user from the called side *registrar* (i.e. utilizes the location service) and then delivers the request to the *intended recipient*. Let us note that the architecture described above is clearly not applicable to MANETs, as they are dynamic networks formed by peer nodes while proxies and registrars are fixed, static and centralized entities. As a result, the SIP protocol, as it is, cannot be deployed in isolated ad-hoc networks [24]. SIP users in MANETs cannot reach other parties, as they do not have support from proxy servers, and cannot be reached by other nodes, as there are no SIP registrars where they can register their contact information. One solution for running SIP in ad-hoc networks [25] could be to elect a node as the registrar in the ad-hoc network; newcomers would retrieve the bindings of the other nodes from it . However, this approach presents several drawbacks in terms of fault-tolerance and scalability. SIP operations in MANETs can be divided into two steps:

1) Discovery of users currently available in the network;
2) Initiating and managing sessions with them.

User discovery can be generic or targeted to find the contact address of a specific individual, e.g. a user in the contact list. Generic discoveries are needed because when a new node joins an ad-hoc network, it usually has no idea of the identities of available users. Initiating an SIP session is not possible if at least the AOR is not known in advance.

2.4 VoIP aware routing protocols, Distributed Rate and Codec adaptation schemes

In literature many VoIP codecs are proposed in order to reduce the impact of the network performance degradation on VoIP calls quality. In [26] the authors propose an algorithm to dynamically adapt the codec on the basis of the perceived call quality degradation. The latter is evaluated through the computation of E-model parameters. Also, in [27] the authors evaluate the impact of codec selection on network performance in an office scenario under VoIP traffic.

In this case, they consider the impact of the number of hops and the number of concurrent flows on the VoIP calls quality under different VoIP codecs. In [28] the authors considered the performance of VoIP traffic under multi-channel MAC and TDMA based MAC exploiting the importance of implementing the Call Admission Control (CAC) in the management of VoIP calls in order to preserve QoS constraints. In [29], the authors emphasize the main factors that affect VoIP traffic performance under dynamic network conditions such as traffic load, network buffer size, scanning Access Point time, transmission rate control to compensate the channel degradation and so on. All these factors can seriously degrade VoIP performance and the capacity of WLAN to support more VoIP calls.

In [30] the authors propose an enhanced AODV protocol called AODVM-ALARM which is able to maintain routes from source to destination and they integrate some control messages that allow a fast reaction to link failures in the routing scheme. They compare the proposed protocol with AODV-UU and AODVM showing how the adoption of a monitoring mechanism to constantly control path and link states can be effective in the VoIP traffic management.

In [31] some mechanisms such as packet aggregation, header compression and adaptive routing are proposed in order to improve VoIP quality and to increase the amount of supportable VoIP traffic, increasing network capacity. They proposed the use of a proactive routing scheme such as DSDV and to extend it with a metric based on the E-model. Differing from this approach, we adopted a reactive and more scalable routing scheme, integrated with SIP protocol, to dynamically manage the codec selection at the application layer. Moreover, we proposed, in addition to the E-model based metric, a flexibility index accounting for traffic distribution to increase the number of VoIP calls accepted by the system while respecting the minimum acceptable MOS value.

In fact, all listed contributions focus on the importance of codec selection to increase the MOS value of the admitted calls under dynamic network conditions. However, only a few of these papers focus on network optimization through traffic distribution of VoIP calls on each critical node under VoIP constraints. Our contribution tries to address this issue and, to the best of our knowledge, there has been no contribution that addressed the joint optimization problem of VoIP calls distribution and dynamic codec selection under QoS constraints such as the minimum MOS value. In the following section, the proposed metric for the routing scheme and the math modeling of the optimization problem is introduced and explained.

3. E-MODEL BASED METRIC AND LEXICOGRAPHIC ORDERING PATH SELECTION

Our proposal is based on an extension of the SIP protocol, due to its robustness and broad diffusion. A distributed scenario has been considered: each node can communicate with its neighbors (a neighbor has a direct radio coverage in the considered environment) and, if a remote communication is requested, a multi-hop path is mandatory, so a key issue is the discovery of neighbor nodes. This problem can be simply avoided by using Hello packets: when a node receives a Hello packet, containing the identity of neighbor nodes, it can easily identify the nodes under its coverage.

To manage a new call properly, the link between source and destination must be bidirectional: forwarding protocols belonging to the OLSR family put tags on the links from a node to its neighbors, in order to know if it is unidirectional or bidirectional. Each node inserts the list of its neighbors in the Hello packet that it is going to send: when a node finds its identity in the received *HELLO* packet, it can tag the link with its neighbor as bidirectional. Note that only bidirectional links are stored in the routing table.

3.1 SIP for MANET

HELLO packets are inserted as headers in the *SIP INFO* packets. As previously discussed, a node does not know the exact (or the best) path to reach a destination; in our proposal, in the SIP INFO request packets, two new headers are present: *Challenge* (to request the elaboration of a path) and *Challenge Response* (to communicate the elaborated path). The *Challenge* packet simply contains the destination node that must be reached and the desired connection parameters. Instead, a *Challenge Response* contains some information about the path that should be considered and the connection parameters that can be guaranteed. Since the source does not know, in advance, the path to the desired destination, the *Challenge* packet is only sent to the neighbor nodes. Each neighbor broadcasts the received *Challenge* to all neighbors and so on. On each hop, the information on the partial path is stored in the *Challenge* packet.

In order to reduce the protocol overhead, *Challenge* packets are broadcast (only towards the neighbors whose link is bidirectional), while *Challenge Responses* are sent only to the source. This way, the choice is not made on the transmitter, but on the receiver node, allowing the system to obtain a notable gain in resource wastage. When a node receives a *Challenge*, it sets a short-duration timer (in order to exclude paths which are too long) and waits for all the *Challenge* packets sent by the same node. When the timer has timed-out, the node answers the best *Challenge* request. This last one will be clarified later on the basis of the proposed metric. After the *challenging* procedure has terminated, the *inviting* phase starts and traditional SIP packets are exchanged.

In order to clarify the behavior of the proposed protocol, in figure 1a a practical example is given: let us assume that node A needs to call node B and all the links in the system are bidirectional. Referring to figure 1a, when the user of node A makes the request, the terminal broadcasts the *Challenge* packet.

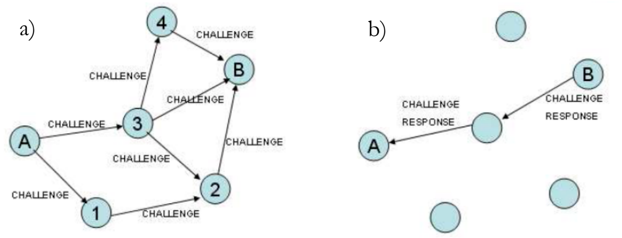


Fig. 1. An example of a simple topology: the challenging phase.

All neighbors will receive and forward (broadcasting) the request. Sooner or later the request will be received by the destination (node B), which will set a timer for the receipt of other *Challenge* packets generated by the same source (node A). When the timer expires, node B will elaborate all received requests and will reply only to the best *Challenge* request (the one belonging to the path with the lowest cost). See figure 1b. It should be noted that in the challenging phase, no intermediary nodes make a back-forward operation. The path having the lowest cost will be computed such as presented in sub-section 3.5.

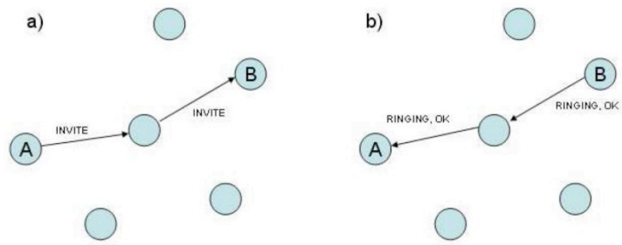


Fig. 2. An example of a simple topology: the inviting phase.

As illustrated in fig.2a, as soon as node A receives the *Challenge Response* packet, it sends the *INVITE* message which contains the desired connection parameters that have been decided in the challenging phase to node B. Node B replies with *Ringing* and *Ok* messages (fig.2b). Finally, the data stream can start after node A sends the *Ack* message. If some replies do not come back to the sender, the request packet is re-forwarded (only by the transmitter node, the intermediary nodes act like traditional stateless proxies).

In our proposal, we hypothesize that information about path resources availability can be stored in the Session Initiation Protocol (SIP) messages; in particular, when a source starts a SIP session, it broadcasts a SIP INFO Challenge (or Request) through wireless relays to destination and each intermediary node stores the information about bandwidth availability in the ongoing message. The Record-Route field [33] of the INFO message can be used to store temporary resource information. Once the destination node received the requests on all the possible paths, containing the encountered bottlenecks in the network, it can choose the best one, unicasting a SIP INFO Response back on the chosen path. At this point the source can invite (make the call) the destination on the best path, following the negotiated parameters (bandwidth, codec, etc.) and when the destination answers the call a SIP OK message is sent back to the source, which acknowledges the call.

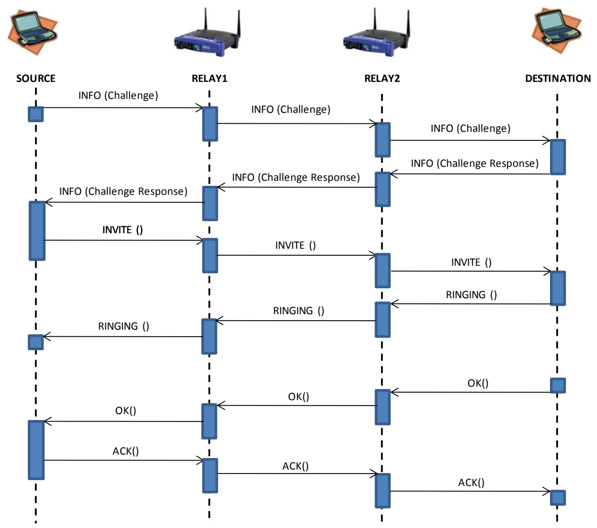


Fig. 3. An example of a simple topology: the inviting phase.

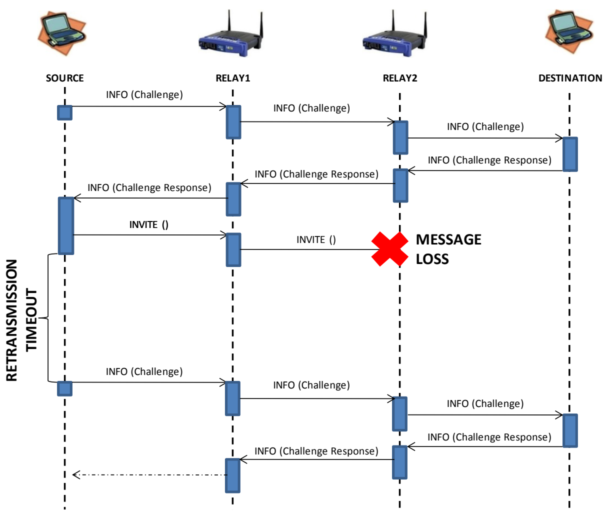


Fig. 4. An example of a simple topology: the inviting phase, in the case of a message loss.

If a message is lost during the signaling phase (in the considered case the SIP Invite message), the communication is temporary stopped, until the source has waited a "retransmission timeout" amount of time. After, the message exchange is started again as in the previous case.

3.2 Codecs Adaptation for mobile nodes

Through SIP protocols it is possible to negotiate and select the most appropriate VoIP codec to adopt in the communication from a source to a destination. In table I the set of possible codecs to apply with their technical characteristics such as data rate supported, payload size and packetization interval are listed. All these features determine different performance of VoIP calls and different situations of degradation sustainable by each flow. On the basis of the E-Model and on the quantitative analysis of VoIP traffic, a different number of calls and QoS constraints can be satisfied.

Table I – Codecs with technical features

Codec	*GSM 6.10*	*G.711*	*G.722*	*G.729*	*G.726 -32*
R	*13.2*	*80*	*48/56 /64*	*8*	*32*
Packetization interval	*20*	*20*	*20*	*30*	*20*
Payload	*33*	*160*	*120/140/160*	*20/24*	*80*

On the basis of the network parameters such as packet error rate, delay jitter and number of admitted flows it is possible to select the best codec in order to respect some QoS constraints and to maximize the number of admitted calls. For example, considering a GSM 6.10 codec, the voice payload is 33 bytes, and 50 packets are generated in each second. After adding the 40 byte IP/UDP/RTP header, the minimum channel capacity is 29.2 Kbps. Also the tolerable packet loss rate depends on the applied codec. For example, codecs with packet loss concealment can tolerate larger packet loss rates. On the basis of this observation, our proposal is the selection of a more appropriate codec and paths (nodes involved in the communications) in order to address QoS degradation in a distributed environment and to increase the number of admitted VoIP calls. In the following sections, a formulation for this problem is presented and a codec and path selection criteria are proposed in order to guarantee minimum QoS satisfaction and to better distribute the VoIP traffic load.

3.3 List of symbols

List of symbols adopted the in the proposed metric and mathematical formulation of the problem are presented below

Table II - List of symbols adopted in the math formulation.

Symbols	
G	*Network graph*
V	*Set of nodes*
E	*Set of links*
N	*Number of nodes*
$d(v_1,v_2)$	*Euclidean distance from* v_1 *to* v_2
DOM	*Set of nodes belonging to a SIP*

	domain
MD	*Maximum coverage distance*
$P(n_1,n_2)$	*Path from* n_1 *to* n_2 *identified by a tuple*
R	*R-factor*
I_s	*Signal to Noise Ratio associated with a typical path*
I_d	*Mouth-to-ear delay*
I_{ef}	*Index of quality degradation associated with the specific equipment*
A	*Expectation factor*
P^{SD}	*Path between S and D identified by a tuple*
L^{SD}	*Set of links associated with the path* P^{SD}
e^{SD}	*Packet error probability related to path* P^{SD}
d^{SD}	*Delay observed on the path* P^{SD}
a^{SD}	*Minimum bandwidth availability related to path* P^{SD}
C^{SD}	*Vector of the load conditions (flexibility index)- Each element of this vector express the load conditions of each node belonging to the path* P^{SD}.
d_{ij}	*Delay associated with the link* $(i,j) \in L^{SD}$
a_{ij}	*Bandwidth availability associated with the link* $(i,j) \in L^{SD}$
$P_e(e(i,j))$	*Error probability associated with the link* $(i,j) \in L^{SD}$
$P_e(i,j)$	*Probability that an error occurs on the link* $(i,j) \in L^{SD}$
U	*Utility function associated with the system*
□ , □	*Vectors of load conditions used in the flexibility index and lexicographic ordering*
f	*flexibility index*
F_R	*Network parameter to be optimized*
MOS_{min}	*Threshold of the minimum acceptable MOS for a call*
r_i	*Rate related to i-th codec*
x_i, y_j	*Discrete decision variables* $\{0,1\}$

3.4 E-model based metric

In this section the criterion used by the system to choose the best path will be described. The traditional choice (made by several routing protocols) is based on the shortest path from source to destination. For the considered systems, this kind of choice is unsuitable, because of the possible existence of bottlenecks in the system, with a consequent reduction in the number of admitted calls and a QoS degradation of the active VoIP flows. As depicted in fig.5, if the choice of the path is based only on the minimum hop number, then node 2 becomes a bottleneck and the whole system may be congested. Thus, a different approach should be used, in order to maximize the number of admitted calls while fitting the QoS constraints.

The introduction of user satisfaction level in the metric is a possible solution to the considered problem, where VoIP calls are present. The *perceived satisfaction level* (expressed as MOS) strictly depends on the *codec* used for voice transmission; better MOS performance requires a higher bit-rate, but if only "good" *codecs* are chosen, a higher bandwidth is required and the number of admitted calls decreases as a consequence. That is to say, users and network have divergent objectives: users want to perceive the best quality and the network tries to accept the highest number of calls. To this purpose, a new metric that does not take only the *hop number* into account needs to be defined ensuring call quality (based on the appropriate codec) and the best path selection to distribute, in an appropriate manner, the data traffic and admit more calls.

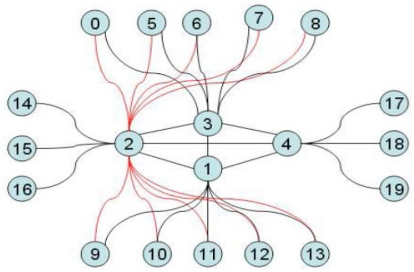


Fig. 5. An example of bottleneck for the traditional approach.

The proposal is now formalized. A network G can be defined as follows:

$$G = \langle V, E \rangle \tag{1}$$

where *V* represents the set of terminals (nodes) that form the network and *E* represents the set of links between nodes. Let *MD* indicate the maximum coverage distance and $d(v_1,v_2)$ indicate the distance between node v_1 and v_2. The set of links can be written as follows:

$$E = \{(v_1, v_2) \mid v_1, v_2 \in V, d(v_1, v_2) \le MD\} \tag{2}$$

Let $DOM \subseteq V$ be the subset of nodes that compose the considered SIP domain (composed of terminals that share the same proxy). For each couple of nodes n_1, n_2 belonging to *DOM* that are endpoints of a requested connection (n_1 calls n_2 for example), it is possible to define a number of paths that interconnect n_1 to n_2, possibly using intermediary nodes (all the nodes in a path are called *relay nodes*). The solution to the original problem is the choice of a path and an appropriate codec. The codec choice can be made by considering the offered MOS, and computing it in real-time through the approach proposed in [17] and extended in [18]. In this way it is possible to follow the *E-Model* to evaluate voice quality for the considered connection between n_1 and n_2 analytically. In particular:

$$R = 100 - I_s - I_d - I_{ef} + A \tag{3}$$

where R is called *R-factor*, I_s is the *Signal to Noise Ratio* (SNR) associated with a typical path in a *Switched Circuit*

Network (SCN), I_d is related to the so-called *mouth-to-ear* delay path, I_{ef} is related to the specific equipment used in terms of codec and packet loss ratio and A is the *Expectation Factor*. The success of the *E-Model* is due to the fact that all the terms are additive and the contributions of delay and packet loss are isolated in the terms I_d and I_{ef} respectively. Note that R assumes its values in the range [0 ÷ 100].

3.5 E-model based Utility function and Mathematical Modeling

Before defining the objective function, the concept of path P must be given in a more detailed way. So, it can be said that a path $P(S,D)$ relating nodes S and D is a tuple:

$$P(S,D) = \left\langle L^{SD}, e^{SD}, d^{SD}, a^{SD}, C^{SD} \right\rangle \tag{4}$$

The first element $L^{SD} \subseteq E$ represents the sequence of links from transmitter node S to receiver node D:

$$L^{SD} = \{(S,i),(i,k),\ldots,(t,j),(j,D)\} \tag{5}$$

The second term e^{SD} represents error probability on the path and it is defined as follows:

$$e^{SD} = \sum_{(i,j)\in L^{SD}} P_e(e|(i,j)) \cdot P_l(i,j) \tag{6}$$

where the term $P_e(e\,|\,(i,j))$ represents the error probability on link (i,j), while the term $P_l((i,j))$ represents the probability to select link (i,j).

The third element d^{SD} represents the delay observed on the entire path, evaluated as the sum of the delays on the links that compose the considered path:

$$d^{SD} = \sum_{(i,j)\in L^{SD}} d_{ij} \tag{7}$$

The term a^{SD} represents the available bandwidth on the entire path, computed as the minimum bandwidth available on the links that compose the path:

$$a^{SD} = \min\{a_{ij} \mid (i,j) \in L^{SD}\} \tag{8}$$

Finally, the term C^{SD} is a vector used to define the flexibility index for route selection. Intuitively this vector stores, for each index i, the number of nodes in the path $P(S,D)$ that are able to admit at most i more calls. It will be formally defined in the next section.

Let $P(S \to D)$ be the set of (tuples indicating the) possible paths from S to D such as defined below:

$$P(S \to D) = \bigcup_{i=1}^{m} \{P_i(S,D)\}, \tag{9}$$

where $m = |P(S \to D)|$. In order to define the *Utility function* (U) associated with the path discovered at the Application layer by the distributed SIP protocol, some decisional variables x_{ij}, y_j need to be defined. The x_{ij} variable is related to the choice of the *i-th* codec on the j-th path:

$$x_{ij} = \begin{cases} 1 & \text{if the } i-th \text{ codec has been chosen on the } j-th \text{ path} \\ 0 & \text{otherwise} \end{cases} \tag{10}$$

The y_j variables indicate the paths to be chosen among the m candidates:

$$y_j = \begin{cases} 1 & \text{if the } j-th \text{ path has been chosen} \\ 0 & \text{otherwise} \end{cases} \tag{11}$$

Now, the utility function U associated with a chosen j-th path from source S to destination D can be defined as:

$$\begin{cases} U_j^{SD} = 94.2 - \displaystyle\sum_{j=0}^{|P(S\to D)|} \left([I_d]_j^{SD} + \sum_{i=0}^{h} [I_{ef}]_{ij} \cdot x_{ij} \right) \cdot y_j \\ \left[I_d\right]_j = 0.024 d_j^{SD} + 0.11 \cdot (d_j^{SD} - 177.3) \cdot H(d_j^{SD} - 177.3) \\ \lfloor I_{ef} \rfloor_j = \gamma_{i1} + \gamma_{i2} \ln(1 + \gamma_{i3} \cdot e_j^{SD}) \end{cases} \tag{12}$$

where $\left[I_d\right]_j$ represents the I_d factor of the *j-th* path and $\lfloor I_{ef} \rfloor_j$ represents the I_{ef} factor of the *j-th* path if the *i-th* codec is chosen. Note that the set of possible paths and the set of possible codecs have cardinalities m and h respectively. The specific numbers such as 94.2, 0.024 and so on are introduced according with other previous valuable contributions such as [32]. This means that U function is computed in a similar way to R-factor expressed in (3).

d_j^{SD} is the one-way delay associated with path j-th and $H(x)$ is the Heavy side function defined as follows:

$$\begin{cases} H(x)=0 & \text{if } x<0 \\ H(x)=1 & \text{if } x\geq 0 \end{cases} \tag{13}$$

Concerning γ_{i1}, γ_{i2}, γ_{i3}, they represent the fitting parameters associated with i-th codec to compute the $\lfloor I_{ef} \rfloor$ value as suggested in [32]. In particular, the expression for I_{ef} associated with path j-th applying the i-th codec under a data loss probability e^{SD} is the following:

$$\lfloor I_{ef} \rfloor_j \approx \gamma_{i1} + \gamma_{i2} \ln(1 + \gamma_{i3} \cdot e_j^{SD}) \tag{14}$$

We can have different e^{SD} for each possible path from S to D because different links with specific error probability are involved in eq.(6), and this justifies the application of different I_{ef} for the considered paths. The same rationale is applied in the one-way delay expressed by $\left[I_d\right]_j$.

3.6 Flexibility index

The proposed route selection is based on what we call path-flexibility index. At each time, for each path $P_j(S,D)$, we maintain a vector C_j^{SD} defined as follows: the i-th position of the vector stores how many nodes in the path are currently able to accept at most i further calls. In particular, the first position, indexed by 0, contains the number of nodes that are not able to accept further calls. Clearly, if the number in this cell is positive, the path $P_j(S,D)$ is already saturated, and useless for future call requests. On the other hand, those paths whose vectors have lower values in the first positions are the most flexible ones, in that have room enough to accept many calls.

An example of C_j^{SD} is given in figure 4: the number in *0-th* position indicates that 5 nodes in this path are not able to admit further calls; similarly, the number in *1-st* position indicates that three nodes on the path will be able to admit one more call, and so on.

0	1		n-1	n
5	3	...	20	0

Fig. 6. An example of vector C_j^{SD}.

Therefore, in the call admission process it is natural to prefer paths (having nodes) that are far from being saturated, that we call flexible paths. Having the vectors described above, this information is readily accessible: the best path will have the lowest value in the first position (saturated nodes) and, among those having the same value in the first cell, will have the lowest value in the second position (quasi-saturated nodes), and so on. Thus, to choose the best path at any time, it is sufficient to maintain these vectors sorted according to the lexicographic order.

Note that the dimension n of the vector may be tuned in order to fit possible time-constraints. The maximum useful dimension is clearly the maximum number of calls that can be accepted by any node in the network. However, in practice even very small values, such as 3 or 4, have proved to be sufficient for the considered application.

	0	1	2	3
j=1	5	3	1	2

	0	1	2	3
j=2	7	2	1	1

	0	1	2	3
j=3	5	1	3	2

Fig. 7. Example of C^{SD} vectors.

In Fig.7 examples of C_j^{SD} structures with a vector size of $n=4$ and $N=11$ number of nodes is shown. The number m of paths from S to D is equal to 3 in this case. Referring to figure 5, following the lexicographic order (field by field), the vector with $j=3$ should be the first, the one with $j=1$ the second and the vector with $j=2$ the third. Formally, a sequence $\sigma=(\sigma_1, \sigma_2, \ldots, \sigma_s)$ comes before a sequence $\tau=(\tau_1, \tau_2, \ldots, \tau_r)$ with $\sigma \neq \tau$ if $\exists\ 1<k\leq s$ so that $\sigma_i=\tau_i$ for $1\leq i<k$ and $\sigma_k < \tau_k$.

Formally, a flexibility index $I_{c,j}$ is defined for each path j and each codec c as follows:

$$I_{c,j} = \sum_{i=0}^{n} v_{c,j}[i]\cdot n^{n-i} \tag{15}$$

where v_{cj} is the vector C_j^{SD} associated with the *j-th* path computed according to the selction of the *c-th* codec. A vector is indeed related to each codec, because on the basis of the selected codec there will be more or less bandwidth occupation on the path. In order to define a normalized index in the range [0;1] where 1 expresses the best flexibility associated with the path and 0 the worst flexibility, it is possible to introduce a flexibility metric

$$F_{c,j} = 1 - \frac{\sum_{i=0}^{|v_j|} v_{c,j}[i]\cdot |v_{c,j}|^{|v_{c,j}|-i}}{N\cdot\left(|v_{c,j}|^{|v_{c,j}|-i}-1\right)} = 1 - \frac{I_{c,j}}{N\cdot\left(|v_{c,j}|^{|v_{c,j}|-i}-1\right)} \quad \text{with}$$

$$0 < f_{c,j} \leq 1 \tag{16}$$

3.7 Optimization problem formulation

Now the constraints can be defined as in eq. (18)-(20). The variable r_i represents the bit-rate of the *i-th* codec, while a_j^{SD} indicates the available bandwidth on the entire path j. The meaning of the first constraint is that the bandwidth required for the new arriving call must be at most equal to the available bandwidth (a_j^{SD}) on the chosen path j-th from source S to the destination D. Let c be the number of codecs available for a mobile node and let m be the number of possible paths from S-D pairs. The constraint on the codec selection of the l-th arriving call for each path j is

$$\sum_{i=0}^{c} x_{ij} r_{ij} \leq a_j^{SD} y_j \qquad \forall j \in P(S \rightarrow D) \tag{18}$$

where variables y_j and x_{ij} are binary decision variables representing the choice of selecting or not the j-th path and the i-th codec for this path, respectively.

In order to impose the choice of only one path for a new call the constraints below can be applied:

$$\sum_{j=0}^{k} y_j = 1 \tag{19}$$

In order to impose the choice of a single codec (the one of the new call which should be admitted) related to the specific path j-th the following constraint can be defined:

$$\sum_{i=0}^{c} x_{ij} = y_j \qquad \forall j \in P(S \to D) \tag{20}$$

To ensure respect of the minimum VoIP QoS for admitted users, it is also required that the MOS is not lower than the minimum threshold (U_{min}):

$$U_j^{SD} \geq U_{\min} \tag{21}$$

Therefore, only codecs that are satisfactory for the users will be chosen, while those that may lead to unacceptable call quality will be discarded.

Among those paths guaranteeing an acceptable level of call quality, the objective function selects one with the best flexibility index. The overall problem formulation is summarized below:

$$\begin{cases} \max_j \min_i F_{i,j} \\ \text{s.t.} \\ \sum_{i=0}^{c} x_{ij} r_{ij} \leq a_j^{SD} y_j & \forall j \in P(S \to D) \\ \sum_{j=0}^{m} y_j = 1 \\ \sum_{i=0}^{c} x_{ij} = 1 & \forall j \in P(S \to D) \\ U_j^{SD} \geq U_{\min} & \forall j \in P(S \to D) \\ y_j \cdot F_{i,j} \cdot x_{ij} \geq F_{\min} \end{cases} \tag{22}$$

where F_{min} is a desired load distribution threshold, to be respected in the network. The strategy of codec selection tries to select the best codec that is able to satisfy the minimum QoS level (U_{min}) and the traffic load distribution threshold F_{min}.

Note that a solution of the above optimization problem may be computed in polynomial time. Indeed, for any path we have to compute a system of linear inequalities, and then we have to sort the flexibility vectors according to the lexicographic order. For completeness, observe that the overall process may be implemented in an efficient way only if the number of possible paths is not very high (and hence the size of the above problem is reasonable). Otherwise, we may think of restricting the paths of interests according to some heuristic criteria (e.g., one may consider only paths whose length is below some fixed threshold).

In fig.8 an example of a simple network with 3 paths from a source S to a destination D is shown. The C^{SD} structure is equal to that presented in fig.5. The path set P is $P = \{P_1, P_2, P_3\}$ with $P_1 = \{S,1,2,D\}$, $P_2 = \{S,3,4,5,D\}$, $P_3 = \{S,6,7,8,9,D\}$, and the flexibility indexes associated with these paths are respectively f_1=0,437, f_2=0,29 and f_3=0,48. On the basis of the lexicographic ordering, the paths are chosen in the following order P_3, P_1 and P_2. This means that considering data traffic and network congestion, path P_3 is preferred even if it is longer than path P_1 because it is formed by mobile nodes with greater capacity. Observe that nodes may not employ their best features to accept more calls, on the basis of the number of current accepted calls and type of applied codecs (low-rate codecs consume little bandwidth, but increase the end-to-end delay reducing the minimum tolerable MOS).

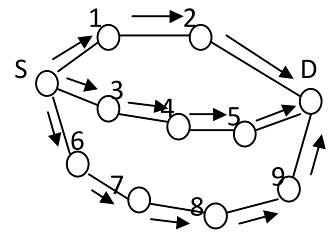


Fig. 8. Network with multiple paths from S to D.

Computation of f_1, f_2 and f_3 with a total number of nodes equal to 11 (N=11) and a flexibility-vector size equal to 4 is presented below. It is assumed to fix a codec c that assures a specific bandwidth usage.

$$f_{c,1} = 1 - \frac{5 \cdot 3^3 + 3 \cdot 3^2 + 1 \cdot 3 + 2}{11 \cdot (3^3 - 1)} = 0{,}416$$

$$f_{c,2} = 1 - \frac{7 \cdot 3^3 + 2 \cdot 3^2 + 1 \cdot 3 + 1}{11 \cdot (3^3 - 1)} = 0{,}262$$

$$f_{c,3} = 1 - \frac{5 \cdot 3^3 + 1 \cdot 3^2 + 3 \cdot 3 + 2}{11 \cdot (3^3 - 1)} = 0.458$$

4. PERFORMANCE EVALUATION

First of all the buffering structure should be described: it is supposed that no packets are dropped for the condition of full buffer (only timeout conditions allow dropping of packets). The kinds of delays considered are: serialization delay (the time needed to insert a packet in the network) and propagation delay (the time needed for the bits to reach the destination). The main contribution to the delay is given by the buffer sojourn time. In the simulator, it is hypothesized that as soon as a packet is inserted into the network, the next one is extracted from the buffer. As described in section III, the size of flexibility vectors may be chosen in order to fit requirements on the protocol performance. In our experiments the evaluation of the flexibility index is based on vectors having size 3. We implemented the SIP protocol over NS2 (application layer) and we changed the network layer in order to consider the flexibility index as routing metric rather than classical minimum hop count metric. In particular two classes (VoIPUDP.cc and VoIPSCTP.cc derived from Traffic class) have been introduced, in order to take into account UDP and SCTP transport layer, as well as the VoIPSIP.cc class (derived from the Agent class). The MobileSIP.cc module has been added, in which it is possible to make some bindings between the OTcl variables and the related C++ ones.

4.1 Topology Analysis and Simulation Scenario

In order to show the usefulness of the proposed metric and protocol, we considered both a synthetic scenario that is critical for *classical* approaches based only on the hop count, and a real-world scenario where our technique is exploited to deal with a port quayside. Here, 50 PDA devices were distributed to workers and five laptops were used as fixed relay nodes. Each mobile node was equipped with the *WiVoIP* protocol, based on the G711 and G729 codecs (both characterized by a constant bitrate of 80kbps and 8kbps respectively).

The topology of the network for the synthetic scenario is shown in Fig.9. Clearly enough, if nodes 0, 5, 6, 7 and 8 make calls with a rate higher than those of nodes 14, 15 and 16, there is a high probability that the latter will be unable to make calls. The topology considered can be subdivided into 5 zones: A, B, C, D and E. It is hypothesized that nodes belonging to zone A (0, 5, 6, 7, 8) make calls only to nodes belonging to zone E (17, 18, 19). Nodes in zone C (1, 2, 3, 4) are relay nodes (neither initiate nor receive calls); nodes in zone D (9, 10, 11, 12, 13) and E (17, 18, 19) only receive calls.

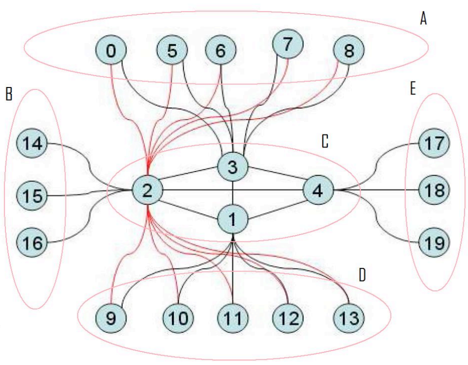


Fig. 9. The critical topology considered.

The main purpose of the first group of simulations is to highlight the advantages of the proposed metric versus the traditional one. The Call Rates (CRs) of zones A and B are λ_A=1/6.5s and λ_B=1/10s respectively. In this way, the nodes in zone A should be faster than those in zone B. The Call Holding Time (CHT) is exponentially distributed, with means μ_A=500s and μ_B=100s. The behavior of the system in function of traffic load has been tested by varying λ_A in the range [0.1, 1] in order to analyze the system behavior under different traffic load: in this way it has been observed how the network saturates in function of the desired F_{min} threshold, as shown in fig.12. Each node is supposed to have a maximum bandwidth of 5.5Mbps with a maximum capacity of 5 simultaneous calls. Each result respects a confidence interval of 95%.

4.2 Simulation Results

Simulation campaigns have been assessed in NS2 and a real test-bed has also been applied in the port area of Gioia Tauro, southern Italy, in order to verify the applicability of our proposal.

The value of γ_i , parameters are presented in the table below as also used in [32].

The minimum MOS value MOS_{min} considered acceptable is 0,70 (70%).

Tab. III – Fitting parameters adopted in simulations

	□ i1	□ i2	□ i3
G.729a	*11*	*40*	*10*
G.711	*0*	*30*	*15*

Performance evaluation has been led out in terms of:

- ***Simultaneous calls***: number of calls supported at the same time.
- ***Call Quality***: it has been evaluated as the perceived R-factor.
- ***Number of admitted calls***: number of call admitted with respect to QoS constraints.
- ***Normalized Overhead***: number of couple Challenge/Response sent on the network divided the number of established calls.

4.2.1 *Simulated Scenario under fixed nodes' topology*

Results on traffic load distribution are shown only for the nodes belonging to zone C, because they are relay nodes and make the system fair.

Fig.10 shows the load level (expressed as simultaneous connections). Node 2 is heavily congested in the classic approach: node 2 belongs to the shortest paths for both zones A and B. Also nodes 1, 3 and 4 managed some active calls: when a node of zone B, rarely, establishes a connection, node 4 is used because it belongs to the shortest path. The load trend on nodes 1 and 3 has a less trivial explanation: due to the high congestion level of node 2, nodes belonging to zone B had few chances to establish calls; node 2 managed only one call from zone B to zone A; the shortest path was congested, so a sub-optimum path was chosen.

Making a comparison with the metric based on the E-Model (indicated with Lexicographic Model – *LexModel* in the figures) it is clear how the load is more balanced among the nodes of the system.

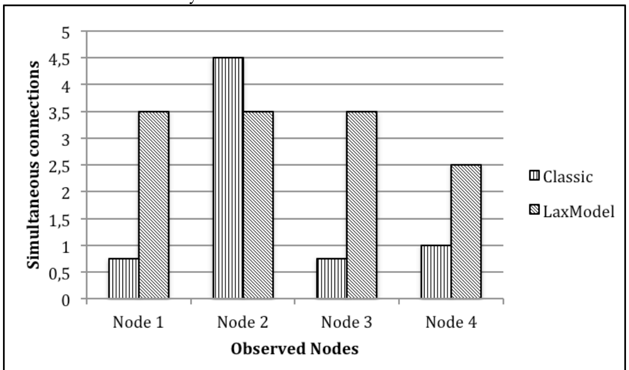


Fig. 10. Traffic Load on central nodes.

Using *lexicographic ordering*, the system noticed that nodes 2 and 4 were crucial as far as network saturation is concerned, so the calls were forwarded through an alternative path (not the shortest one) composed of node 1 and node 3. The unbalanced load is not the only problem introduced by the classical approach: also fairness is not respected, because, as depicted in figure 8, nodes belonging to zone B are not able to make calls if the traditional metric is used. With the *LexModel,* this trouble does not occur, in fact each node is able to make or receive calls. We say that the fairness criterion is guaranteed.

The MOS value (expressed as R-factor) was evaluated for both metrics: a value of 82.5 was obtained for the *LexModel,* while a value of 77.86 was obtained for the traditional metric. Both values are satisfactory.

As expected, the synthetic topology is critical for the classical approach. Indeed, in this network few nodes are required to distribute data traffic in order to manage more calls, and they can easily be overloaded, if some load-balancing and traffic distribution based on the R-factor is applied.

In fig.12, it is shown the graph of R-factor evaluation for Lexmodel algorithm under different minimum threshold F_{min} related to the load distribution to apply in the network. In this case, it is evident as a lower threshold assures an higher saturation point decreasing the quality of the VoIP calls. On the other hand, an higher threshold reduces the number of admitted calls because more performing codecs are selected.

In fig.13, the normalized protocol overhead computed as the number of Challenge/Response packets sent on the network on the number of admitted calls. This means that in the LexModel, when a traffic load of 10% is observed on the network and the overhead is 1, to find a path and establish a call we need of 1 couple of packets (Challenge/Response). On the contrary. for the classic model, the overhead is higher because can happen that we are not able to find an available path during the route discovery because not VoIP aware metric is applied. Thus, more trial need to be performed. This trend worses for higher mobility speed where the specific metric is more performing in comparison with the classic model.

However, LexModel degrades its performance in terms of overhead for increasing traffic because more calls generate separate Challenge/Response packets to establish different VoIP sessions. This does not mean that the protocol is not scalable but that if an on-demand scheme is applied. its benefits are reduced when the traffic load is very heavy due to the high number of discovery to be performed.

A proactive scheme could be applied to maintain in-advance the topology and QoS states of the network.

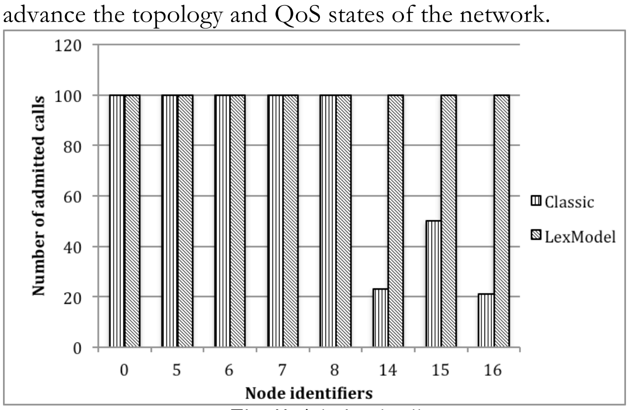


Fig. 11. Admitted calls.

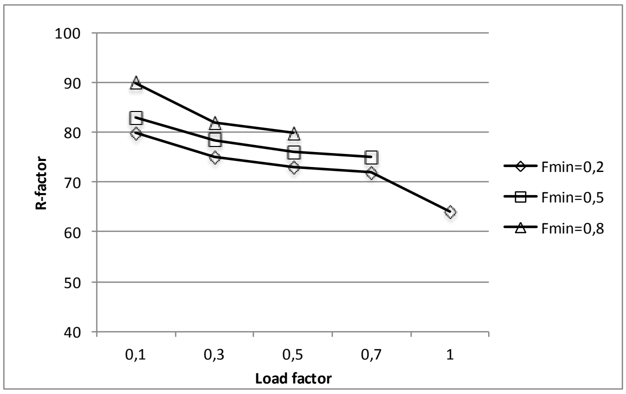


Fig.12. R-factor vs increasing traffic load with different minimum load distribution factor.

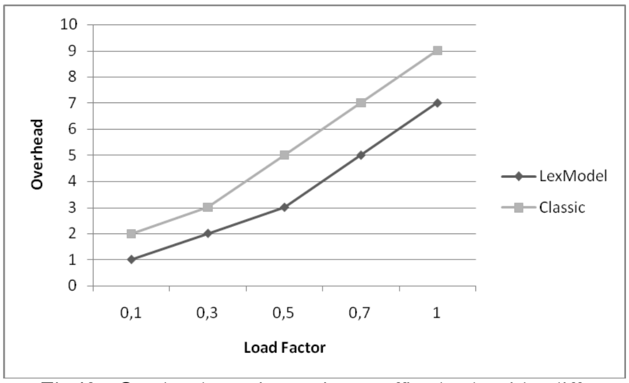


Fig.13. Overhead vs increasing traffic load with different minimum load distribution factor.

However, in this last case, a price is paid under low traffic conditions. A deep evaluation of the overhead optimization under the very dense scenario is out of the scope of the following work. A starting point where to address future contributions can be found in [34][35].

4.2.2 *Simulated scenario under nodes' mobility*

The second simulation campaign considered a network of 50 nodes where a mobility with speed in the range [0-10m/s] and a traffic load of r0=0,8 and r0=0,6 are considered.

In fig.14 it is possible to see as the Lexmodel outperforms the classic model. In particular Lexmodel is able to respect the QoS constraint expressed in terms of R-factor. The classic protocol with the addition of a basic CAC limit the number of accepted calls and this allow a better respect of QoS limits. However, the perceived QoS is greater with our proposal because the joint effect to limit the number of calls and to distribute the VoIP traffic leads to a better QoS degree and an higher number of admitted calls if the QoS constraint is pre-fixed. In fig.15, the Classic model makes use of CAC to limit the number of calls respecting QoS metric. However, our proposal is able to admit more calls respecting the min MOS (or R-factor) and distributing better the data traffic. This means that the network saturation point is reached later such as shown in Fig.15 where the number of Lexmodel is greater than Classic approach.

For all protocols we assist to a general QoS degrading of the calls because the node mobility and the reduced route stability determines a severe degradation. This means that R-factor perceived is significantly reduced and a higher number of calls is reduced is the mobility is higher than 4m/s. Also if the performance are degraded, flexibility index and Lex model are useful to reduce the degradation and the rejection of calls.

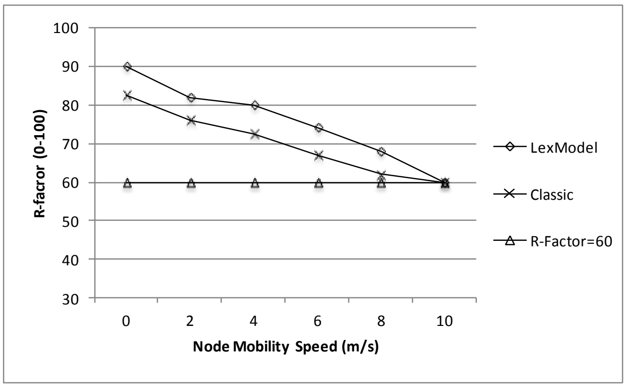


Fig.14. R-factor evaluation for calls under increasing mobility speeds and with different protocols.

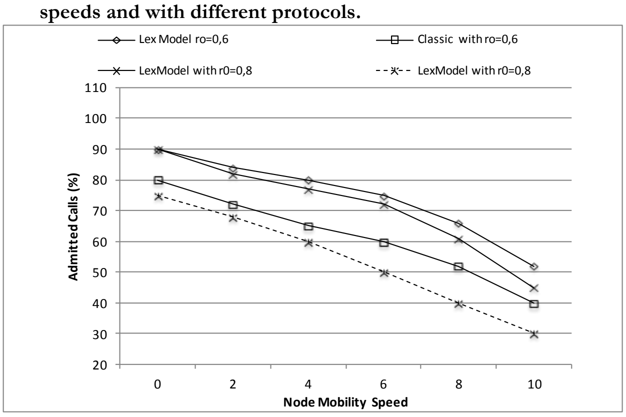


Fig.15. Number of admitted calls vs increasing mobility speeds and with different protocols.

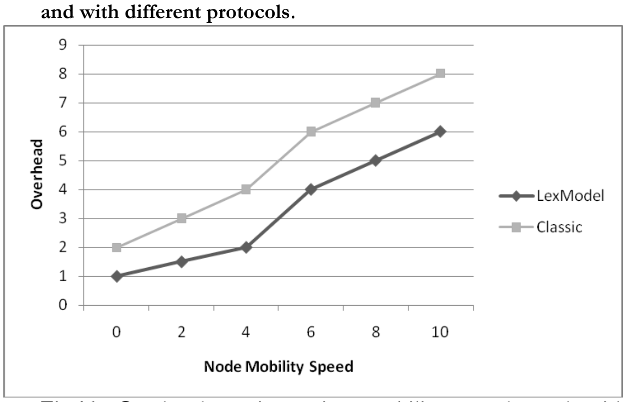


Fig.16. Overhead vs increasing mobility speeds and with different protocols.

In fig.16, as it is expected, normalized overhead increases due to the route breakage that determines new Challenge/Response packets to be sent on the network to establish again the call. Such as recalled in the previous sub-section, other routing schemes could be applied with our metrics and with the same optimization problem to evaluate the scalability issues in terms of load, mobility and number of nodes (sparse and dense networks).

4.2.3 Real-world scenario

The third campaign of simulations was carried out considering the real-world scenario, in order to demonstrate that a more effective metric is suitable also in real cases and not only in synthetic critical ones. The considered scenario consists of a port area where there are some ships, containers and goods that need to be managed in some way. Each job is assigned to a team of workers, who communicate using smart-phones or PDAs with VoIP software installed. Each team has a leader, who acts as coordinator and who can communicate with the port headquarters to inform them of some critical situations (emergencies, etc.). Only the headquarters have fixed stations that act as supervisors. There are many supervisors that monitor more than one group. The multi-hop protocol runs on each terminal and if a higher extension of the port area is considered (about 6 km^2), the different teams are, potentially, isolated: due to the insufficient power of the mobile devices, each member of a team can communicate with each other, but members of different teams cannot communicate with one another directly. To solve this problem, some stations (inexpensive terminals) are fixed in some strategic points, in order to guarantee a minimum coverage of the interested space.

Fig.17 shows the considered scenario for the third simulation campaign; circles represent fixed stations, while clouds represent the different operating teams and the headquarters. In this scenario there can be an unbalanced load but the fairness criterion must be satisfied, because each call that is made by a supervisor of a team to the headquarters should always be accepted (the calls have high priority); each node of a working-team belongs to the direct coverage area of the associated fixed point: each call of a worker directed to another worker of the same team is able to follow a short path composed of three nodes (source, fixed node and destination). Each *cloud* (working team) is composed of ten workers, and two of them are elected as *supervisors* (one is the *real supervisor* and the other one a *quality controller*).

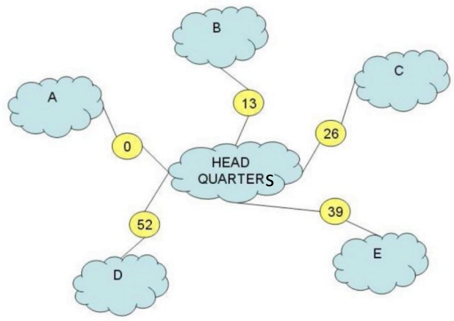


Fig. 17. A real application scenario.

Calls are generated randomly and the caller always tries to contact an endpoint that is at least two hops away from him (in this way, the total path is thus composed by four nodes). Calls between workers of the same team are made frequently with short duration (the chosen parameters are Poissonian CR with $\lambda_{TEAM}=1/6s$ and exponential CHT with μ_{TEAM}=150s), while calls between supervisors and headquarters are not frequent and a longer communication time is applied (the chosen parameters are Poissonian CR with $\lambda_{SUPER}=1/20$s and exponential CHT with μ_{SUPER}=300s). The fixed nodes 0, 13, 26, 39 and 52 can handle only two simultaneous calls. Ideally, each supervisor should always be able to make calls and the fixed node should not be used to manage calls between workers of the same team. Therefore, the protocol should recognize that fixed nodes are critical for the fairness of the system and avoid using them to manage calls among workers of the same team. In the simulation results the load on the nodes is not shown, because in this scenario we are not interested in load-balancing issues.

Fig.18 shows that, if the traditional metric is used, for all the working teams the percentage of admitted calls is very low (a mean value of 25%) with high variance (due to the random generation of links for calls between workers of the same team, alternative paths that do not use the fixed node can be found).

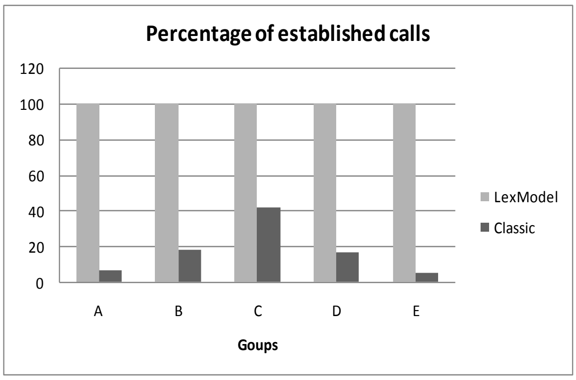


Fig. 18. The trend of admitted calls.

When the proposed metric is used (based on the E-Model) the admission percentage is always 100%: No chosen path blocks any sector of the system, and the fixed nodes are used only when necessary (calls from supervisors to headquarters). Also in this case the R-factor is considered: for the traditional metric a value of 85.78 is obtained, while for the proposed *LexModel* metric a value of 87.12 is observed: with the classical approach, for a couple of calls between workers of the same team, on average, the fixed node is used, while for the other calls longer paths are used; using the proposed metric, team calls are forwarded on longer paths, while for calls directed to the headquarters fixed nodes are used. So the obtained value of *R*-factor is similar. Note that if the fixed nodes are substituted with more powerful ones (able to manage up to ten simultaneous calls) the classical metric will not show its limitations, but only a worse *R*-factor, due to the overload of the fixed stations.

5 Conclusions

In this paper a novel metric based on an objective measure of the QoS of VoIP calls has been proposed. It allows the use of VoIP services in distributed wireless networks and hostile environments. In particular, it has been shown how the use of the E-Model and of a flexibility index based on a lexicographic ordering allow both user and network requests to be met, typically these being difficult to manage together. The proposed metric reduces the probability of a block in the network to the minimum and increases the number of admitted calls in the system. The soundness of the proposal has also been verified by analyzing the average MOS values, which have always been higher than the one obtained with a classical approach, based on the hop count. Simulation campaigns were assessed in NS2 and a real test-bed was also applied in the port area of Gioia Tauro, southern Italy, in order to verify the applicability of our proposal.

Acknowledgment

This work was supported by the AUTOMA Project established under the PON program in the port area of Gioia Tauro, Reggio Calabria, Italy.

References

[1] H.Kazemitabar et al., "A comprehensive review on VoIP over Wireless LAN networks," in *Computer Science Letters*, Vol.2, Seèt.2010.

[2] "The E-Model, R Factor and MOS, Overview", http://www.sageinst.com/downloads/960B/EModel_wp.pdf, Dec. 2003.

[3] Rosenberg et al. SIP: Session initiation protocol. RFC 3261, IETF, June 2002.

[4] E. Bernex, A. Gatineau, "Quality of service in VoIP environments", White Paper, www.neotip.com.

[5] H. M. Chong, H. S. Matthews, "Comparative Analysis of Traditional Telephone and Voice-over-Internet Protocol (VoIP) Systems", IEEE ISEE 2004.

[6] Lijing Ding, Rafik A. Goubran, "Assessment of effects of packet loss of speech quality in VoIP", in *2nd IEEE Int. Workshop on Haptic, Audio and Visual Environments and Their Applications, HAVE 2003,* 20-21 Sept., 2003, pp.49-54.

[7] C.Perkina, E.B.Royer, S.Das, "Ad hoc On-Demand Distance Vector (AODV) Routing," in *RFC 3561*, IETF org.

[8]T.Clausen, P.Jacquet, Optimized Link State Routing (OLSR), *tech. specifications in IETF RFC 3626*, 2003.

[9] F.De Rango, et al., "OLSR vs DSR: A comparative analysis of proactive and reactive mechanisms from an energetic point of view in Wireless Ad HocNetworks," *Computer Communication Journal*, Elsevier, Oct.2008, pp. 3843-3854.

[10] F.De Rango, M.Tropea, P.Fazio, S.Marano, "Overview on VoIP: Subjective and Objective Measurement Methods," in *International Journal of Computer Science and Network Security (IJCSNS)*, Vol.6, No.1, Nov.2006.

[11] A. Lakaniemi, J. Rosti, V. I. Raisanen, "Subjective VoIP speech quality evaluation based on network measurements", in *IEEE ICC 2001*, 11-14 June, Helsinki, pp.748-752.

[12] ITU-T Recommendation P.800, "Methods for Subjective Determination of Transmission Quality", 1996.

[13] J. Anderson, "Methods for Measuring Perceptual Speech Quality", White Paper, Agilent Technologies, Oct. 2001.

[14] F. Hammer, P: Reichi, T. Ziegler, "Where Packet Traces Meet Speech Samples: An Instrumental Approach to Perceptual QoS Evaluation of VoIP", in *12th IEEE International Workshop on Quality of Service (IWQOS 2004),* 7-9 June, 2004, pp.273-280.

[15] ITU-T Rec. G.107, "The E-model, a computational model for use in transmission planning", Mar. 2005.

[16] Rafik A. Goubran Lijing Ding, "Speech quality prediction in voip using the extended e-model," Globecom 2003, 1-5 Dec., vol.7, pp. 3974-3978.

[17] www.itu.int/ITU-/studygroups/com12/emodelv1/index.htm.

[18] www.psytechnics.com.

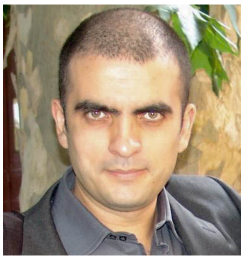
[19] ITU-T Recommendation P.862, "Perceptual Evaluation of Speech Quality (PESQ), an Objective Method for End-to-End Speech Quality Assessment of Narrowband Telephone Networks and Speech Codecs", Feb. 2001.

[20] "PESQ: An Introduction", WP, Psytechnics Sep. 2001.

[21] A.E. Conway, "Output-Based Method of Applying PESQ to Measure the Perceptual Quality of Framed Speech Signals", in

IEEE Wireless Communications and Networking Conference (WCNC 2004), 21-25 March, 2004, vol.4, pp.2521-2526.
[22] "PESQ: Measuring speech quality over networks, as the customers hear it", ", *White Paper*, Psytechnics 2001.
[23] Alan B. Johnston. Sip: Understanding the session initiation protocol, Artech House Publishers, 2009.
[24] T. Moors, M. Veeraghavan, N. Cocker, "Support of voice services in ieee 802.11 wireless lans,". *Technical report*, Polytechnic University, 2000.
[25]P.Studei, M.Bihr, A.Remund, G.Alonso, "SIPHoc: Efficient SIP Middleware for Ad Hoc Networks," in *Lecture Notes in Computer Science*, 2007, Vol.4834, pp.60-79.
[26] A.Sfairopoulou, C.Macian, B.Bellata, "VoIP codec Adpatation Algorithm in Multirate 802.11 WLANs: Distributed vs Centralised Performance Comparison," *Dependable Adaptable Network Services*, pp.52-61, Springer, 2007.
[27] M. Cardenete-Suriol, M. Portoles-Comeras, J. Mangues-Bafalluy, "Performance issues for voip call routing in a hybrid ad-hoc office environment," in IST Summit, 2007.
[28] A. Chan, S. Chang Liew, "Performance of VoIP over Multiple Co-Located IEEE 802.11 Wireless LANs", *IEEE Transactions on Mobile Computing*, Volume 8, Issue 8, Aug. 2009 Page(s):1063 – 1076.
[29] S. Shin, H. Schulzrinne, "Measurement and Analysis of the VoIP Capacity in IEEE 802.11 WLAN", *IEEE Transactions on Mobile Computing*, Volume 8, Issue 9, Sept. 2009, Page(s):1265 – 1279.
[30] S.Ganguly et al., "Performance Optimization for Deploying VoIP Services in Mesh Networks," in *IEEE International Journal on Selected Area on Communications (JSAC)*, 4-th quarter, 2006.
[31] A.S:Otero, M.Atiquzzaman, "Adaptive Localised Active Route Maintanance Mechanism to Improve Performance of VoIP over Ad Hoc Networks," in *Journal of Communications*, Vol.6, No.1, Feb.2011, pp.68-78.
[32] R.G.Cole, J.H.Rosenbluth, "Voice over IP Performance Monitoring," in *ACM Computer Communication Review*, Vol.31, Issue 2, April 2001, pp.9-24.
[33] S. Donovan, IETF RFC 2976 The SIP INFO Method, Oct.2000.
[34] Biao Zhou, Yeng-Zhong Lee, Mario Gerla, Floriano De Rango, "Geo-LANMAR: a scalable routing protocol for ad hoc networks with group motion," in Wireless Communications and Mobile Computing 6: 989-1002 (2006).
[35] Santivanez CA, Ramanathan R, Stavrakakis I, "Making Link-State Routing Scale for Ad Hoc Networks," in Proceedings of the 2nd ACM international symposium on Mobile ad hoc network-ing & computing (MobiHoc 2001); Long Beach, CA, USA, 2001, 22–32.

Floriano De Rango graduated in Computer Science in October 2000, and a Ph.D. in electronics and telecommunications engineering in January 2005, both at the University of Calabria, Italy. From January 2000 to October 2000 he worked in the Telecom Research LAB C.S.E.L.T. in Turin with a scholarship. From March 2004 to November 2004 he was visiting researcher at the University of California at Los Angeles (UCLA). He is now Assistant Professor at D.E.I.S. Department, University of Calabria. He was recipient of the Young Researcher Award in 2007 and is reviewer TPC member for many International Conferences and reviewer for many journals such as IEEE Communication Letters, JSAC, IEEE Trans.on Vehicular Technology, Computer Communication, Eurasip JWCN, WINET etc. His interests include Satellite networks, IP QoS architectures, Adaptive Wireless Networks, Ad Hoc Networks and Pervasive Computing. In 2009 he founded a spin-off company (Spintel s.r.l.) in the smart control tools design field for energy efficiency for Smart Building and Cities and Internet of Things (IoT) that won some awards in International Competitions such as the Intel Business Challenge award. His company has been ranked in the first 20 innovative start-up in Europe (IBCE 2013). Homepage: http://culture.deis.unical.it/derango; e-mail: derango@deis.unical.it.



Peppino Fazio graduated in Computer Science in May 2004. Since November 2004 he has been a Ph.D student in Electronics and Communications at the University of Calabria and obtained his Ph.D. in January 2008; at the moment he is a research fellow at DEIS Department of University of Calabria and is collaborating with the "Universidad Politecnica de Valencia". His research interests include mobile communication networks, QoS architectures and interworking wireless and wired networks, mobility modeling for WLAN environments and mobility analysis for prediction purposes.

Francesco Scarcello received the Ph.D. degree in Computer Science from the University of Calabria in 1997. He is a professor of computer science (SSD ING-INF/05) at the University of Calabria. His research interests are computational complexity, game theory, graph and hypergraph theory, constraint satisfaction, logic programming, knowledge representation, non-monotonic reasoning, and database theory. He has extensively published in all these areas in leading conferences and journals. Professor Scarcello holds his current position since 2001. Before that, he was recipient of two grants from the Italian National Research Council (CNR) until 1999, when he became an assistant professor at the University of Calabria. Also, during these years, he has been visiting the Department of Information Systems at the Vienna University of Technology.
Professor Scarcello serves on program committees and as a reviewer for many international conferences and journals. He is co-recipient of the 2008 IJCAIJAIR Best Paper Prize, awarded to an outstanding paper published in the Journal of Artificial Intelligence Research in the preceding five calendar years.
He is co-recipient of the 2009 ACM PODS Alberto O. Mendelzon Test-of-Time Award, awarded every year to a paper published in the proceedings of the ACM Symposium on Principles of Database Systems (PODS) ten years prior that had the most impact in terms of research, methodology, or transfer to practice over the intervening decade.

Francesco Conte graduated in Computer Science in July 2009. Since September 2009 he has been consultant in design and development of enterprise solutions for office automation and domotics. At the moment he collaborates with "Banca d'Italia" in designing solutions for fraud detection and statistical data elaboration. His additional interests are social graph, text mining, agile techniques and Java EE.