Modeling IP Telephony Call with Simple Algebraic Relations

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***Abstract*—During the evolution of telephony, many models have been proposed. However the models usually consider only certain aspects of the telephony network. This paper presents an approach to the use of algebraic relations in mathematical modeling of IP telephony networks. Three algebraic relations that represent the state of the network are defined. An interesting property of telephone network is demonstrated, that a telephony session or "call" can be interpreted as the class of equivalence of one of relations. Session life cycle is explained in the view of defined relations and the time dimension of events is briefly discussed. The relations presented here are foundations for a more comprehensive and usable model. The directions for future work are given.**

Keywords- IP telephony; telephone call; VoIP session; Session Initiation Protocol (SIP).

I. INTRODUCTION

Traditional telephony networks have been extensively researched in the period of more than a century. Majority of researches consider formal modeling of only certain aspects of the network functionality. The telephone exchange has been modeled using queuing theory and different parameters, such as call waiting time, probability of blocking, etc. that are estimated.

The IP telephony appeared in the last decade of the XX century. Two most important signaling protocols for IP telephony are Session Initiation Protocol (SIP) [1] and H.323 [2]. Again, queuing theory is used for estimating parameters such as end-to-end Voice-over-IP (VoIP) connection delay, see [3] for a recent work. In this paper, the approach to model mathematically several most important elements of the functionality of the IP telephony network is presented. The model is proposed having in mind the SIP protocol, as it is dominant today.

In the majority of available public documents in the telephony area, the call is not defined, but it is assumed that a reader already knows what the call is. Starting from the structure of the telephony network, and analyzing some aspects of network dynamics, we will try to shed new light on the concept of a telephone call.

The foundation of the telephony network is the set of network nodes (N), which are either endpoints (EP) available to users, or infrastructure points (IP) that make the infrastructure of the network. There are several types of IP:

- routing points (RP)
- conference points (CP). Those are conference servers and media mixers used for the support of conference feature with respect to signaling and media streaming, respectively.

Thus,

$$IP = RP \cup CP, \tag{1}$$

$$EP \cap IP = \varnothing,$$

$$N = EP \cup IP.$$

Nodes are connected by links. The set of links can be defined as

$$L=\{(x,y)|(x,y) \in EPxIP \vee (x,y)\in IPxIP \vee (x,y) \in IPxEP\}, \tag{2}$$

L is a symmetric relation – we consider links that ensure bidirectional communication.

The mapping of the structure of a so-called pre-IMS IP telephony network (still much in use today) to this model is straight forward. On the other hand, the IP telephony network with the IP multimedia Subsystem IMS [4] environment has richer structure with a larger number of different functional elements, but their mapping to RP and CP is possible. The fact that Call Session Control Function (CSCF) elements are realized as SIP servers makes the mapping straight forward.

The traditional telephony is limited with the fact that calls cannot be routed based only on signaling information while call processing applications require access to media stream, as well. This results in so called “tromboning issue” that has been solved with the advent of VoIP in which the signaling and media stream path can be fully separated [5].

The section II contains an overview of some related work. In section III, the mathematical relations that are the foundation of our model are presented. In section IV, the session life cycle is discussed in the view of relations defined in section III. The section V contains concluding remarks and directions for future work.

II. RELATED WORK

There are many approaches to formal modeling of communication protocols (as a more general class containing

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telephony communications). The use of Petri nets dates back several decades ago. In the field of telephony communications, the ITU-T directed efforts for a long time to establish software development methods supported by formal models. Especially in the area of feature interaction, several approaches to formal modeling which provided rapid detection of unwanted feature interactions were proposed [6]. A more recent comprehensive work on feature interaction problem is presented in [14]. Some other recent approaches are in [16] and [17]. An example of the use of Petri nets in modeling telephony networks is given in [7]. In the packet based telephony, there is also work in the development of services supported by formal models and a segment of that work is presented in [8]. In [19], authors applied formal modeling and verification of several fixes for privacy issues in 3G. In [20], authors present network topologies used by several popular chat applications. The structural model presented in this paper suits the topology used by Google+, while it differs from peer-to-peer and hybrid topologies of iChat and Skype, respectively.

With respect to modeling the structure of large IP telephony networks, important sources are documents published by IP telephony vendors. Those, usually corporate networks, are covered as in [12], where single and multisite models for corporate IP telephony are presented. The taxonomy in [13] is a bit more elaborate and it distinguishes single site, centralized and distributed call processing model, but it still covers only a corporate network case. On the other hand, an example of research in the structure of network is [18] where authors analyze clustering of bipartite Internet-to-telephone network ItPBN (IP telephone calls from Internet to traditional telephone network) and show that it has power law incoming/outgoing degree distribution. The ItPBN has one giant component and a large number of satellite components. The size of satellite components follows the power law.

In the first decade of XXI century, the SIP protocol gradually claimed dominant place as VoIP signaling protocol, leaving behind H.323. A proposal for a new VoIP signaling protocol, that overcomes some deficiencies of the SIP is presented in [15].

One of the trends in IP telephony is development of peer-to-peer (P2P) IP telephony systems. Some existing commercial systems already deploy P2P or hybrid architectures (see [20]). P2P SIP is an open technology based on this concept, see [21] as an example of recent work in this area. Ref [22] presents a P2P IP telephony system for wireless ad-hoc networks. Ref [23] presents a strategy to form the overlay network of P2P IP system.

It is stated [10] that one of drawbacks of traditional telephony models (Integrated Services Digital Network - ISDN, Intelligent Network - IN being good examples) is that they are based on the notion of a call. The call is described as an obscure and difficult notion [10]. Distributed Feature Composition architecture [9] that should solve that problem was proposed in late 90s, but did not enter in wide use. In our opinion, this property of the notion of the call stems from the fact that it is perceived both from the standpoint of an end user of telephony services and from the standpoint of an engineer developing services - and these two projections sometimes collide. This paper is, among other things, an attempt to say what the call is, but in terms of mathematical reasoning.

III. MATHEMATICAL RELATIONS

The state of network is described by a Connectivity State (CS) set. This set contains the following three most important relations (physically connectable - PhyC, connected - Conn, and negotiating - Nego). Nego relation covers both negotiation of session parameters and session establishment operation.

$$CS = \{PhyC, Conn, Nego\} \tag{3}$$

The PhyC relation represents current infrastructure state. It is the set of available communication paths. The PhyC relation is the subset of the set EPxEP. We consider this relation to be symmetric and transitive, but we note that in special cases this assumption may not hold - depending on the configuration of routing tables and polices. If there is an active link between *x* and *y*, then both (x,y) and (y,x) are members of the PhyC. Also, if there are links that connect *x* with *y*, and *y* with *z*, this means that $(x,z) \in$ PhyC. It can be assumed that the PhyC is reflexive. In this case, the class of equivalence of *x* that is the member of EP represents the connectivity potential of that endpoint. The *y* that is the member of EP and the member of the class of equivalence of *x* is an endpoint that is reachable from *x*. In a normal operation, there is only one class of equivalence - any endpoint is reachable from any other endpoint.

Conn and Nego relations are the subset of the set $\cup N^i$, i=2,...,n, where n is the greatest number of users that can participate in one telephony session - in that network. Or, more strictly limited,

$$Nego, Conn \subset \{EPxEP \cup (EP^i xCP), (i=1,...,n-1)\} \tag{4}$$

Neither of the relations is intrinsically reflexive. Both are symmetric and transitive. If $x,y \in N$ and *x* is connected to *y* by a telephony session, then *y* is connected to *x* too - because of characteristics of the telephony session. Also, because of characteristics of the telephony conference, if $x,y,z \in N$ and *x* is connected to *y* by the telephony session, and *y* is connected to *z*, then *x* is connected to *z*, too. In our model, the telephony session is a SIP session.

Again, because of characteristics of the telephony session, if *x* is negotiating with *y*, then *y* is negotiating with *x*, too. Transitivity of the Nego relation is not so obvious, but if we observe $x, z \in EP$ and $y \in CP$, it can be said that if both *x* and z are negotiating with *y*, then *y* is negotiating with *z* (because of symmetricity), and it can be said that implicitly, *x* is negotiating with *z*, too. For *y* which is not a conference point, this does not hold.

It can be seen that

$$((\forall\ (x_1,...,x_{n-1}) \in EP^{n-1},\ \forall\ x_n \in CP\ )\ |\ ((x_1,...,x_n) \in Conn) \vee$$

$$((x_1,...,x_n) \in Nego)) =>$$

$$(\forall (x_i,x_j)\ |\ 1 \leq i,j \wedge\ i,j \leq n => (xi,xj) \in PhyC) \tag{5}$$

While the PhyC is a relation that represents connectivity, and in terms of network technology it belongs to so called communications oriented layers of ISO OSI [11] protocol stack (physical, data link, network and transport layers), Conn and Nego are about signaling, and belong to so called application oriented layers upper layers of ISO OSI stack (session, presentation and application layers), or the application layer in TCP/IP [11] stack.
For each n-tuple that is negotiating or connected at the moment, there has to exist a network path connecting each of the members of the n-tuple $(x_1,...,x_n)$.

$$(\forall\ (x_1,...,x_n)\in Conn)\ \ (\forall\ (x_i,x_j),\ 1\leq i\leq n,\ 1\leq j\leq n)$$

$$(\exists\ R\subset RP)\ (\exists\ r_k,\ r_l\in R\ |\ (x_i,r_k)\in L\wedge(x_j,r_l)\in L)$$

$$(\exists\ f:N\text{->}R\cup\{x_i,x_j\}|\ f(1)=x_i\wedge f(|R\cup\{x_i,x_j\}|)=x_j\ \wedge$$

$$(\forall\ i\in N)\ \ (i<|R\cup\{x_i,x_j\}|\ =>(f(i),f(i+1))\in L)) \tag{6}$$

The same is true for $(\forall\ (x_1,...,x_n)\in Nego)$.

Where network path is a set of links connecting nodes belonging to R, and it is determined by relation f. For that reason, for each n-tuple that is an element of Conn or Nego, there is an m-tuple (n<=m) containing both the elements of R and the elements of the n-tuple. During the session establishment, certain resources are allocated at each member of the m-tuple. Upon the successful session establishment, depending on the type of session (i.e. use of Route headers [1] in SIP), the resources allocated at the elements of R are freed. In certain applications (especially those based on use of Back-to-Back User Agents [1]) it is required that routing points stay in the routing path during the whole session duration.

We can postulate that Conn relation is reflexive, because logically users at any endpoint can hear themselves during the session (echo functionality). In that case, Conn is a relation of equivalence. A telephone call in that case is an equivalence class of relation Conn.

If we postulate, a bit artificially though, that Nego is reflexive too, in which case that relation is a relation of equivalence too, it can be said that a class of equivalence of the Nego relation is a telephony session candidate.

The network load (in the sense of a set of ongoing network activities) can be projected into two functions of time:

$$LoadConn:\ t\rightarrow Conn;\quad LoadNego: t\rightarrow Nego. \tag{7}$$

One important characteristic of the network can be noted as:

$$(\forall t_2)(Time(t_2))(\ \forall x,y\in EP)((x,y)\ \in LoadConn(t_2))\ =>$$

$$(\exists t_1)(Time(t_1))\ (t_1<t_2)\ ((x,y)\ \in LoadNego(t_1) \tag{8}$$

Or, in other words, only those nodes that in the earlier moment of time belonged to the class of equivalence of x in relation Nego can later belong to the class of equivalence of x in relation Conn.

We remark though, that not all negotiations result in session establishment, which means that not all members of the class equivalence of x in relation Nego will become members of the equivalence class of x in relation Conn.

IV. Session life-cycle

Presented relations make a model of associations between nodes of the network during its operation. However, they tell little about events that occur during one session.

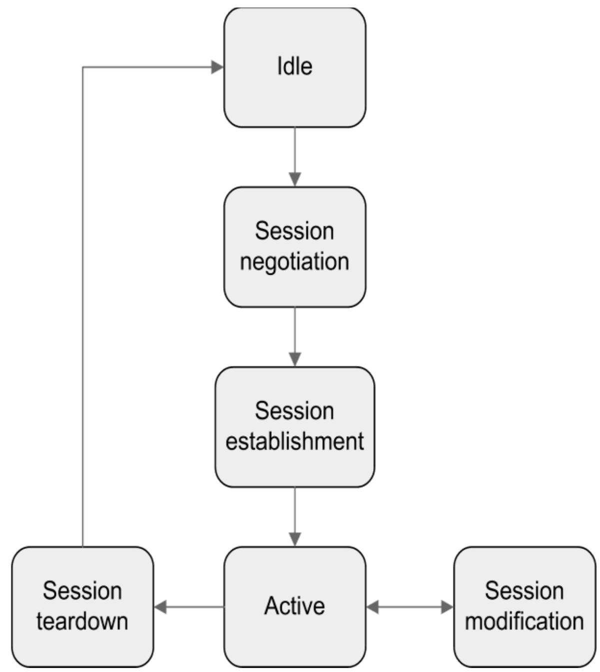

Figure 1. The session life-cycle.

Table 1 describes the most important states in the session life-cycle.

The states given in the table are additionally described with the following statements:

- After the session modification, *ep* returns to active state.
- During the session teardown, *ep* is deleted from the Conn relation.
- After the session teardown, *ep* returns to the idle state.

In this paper, we consider stable state to be the state in which the *ep* can stay unlimited time - in general case. Duration of transitional states is controlled by timeout controls specified by the telephony protocol, and this duration is on the average small in the sense of human perception, and compared to the average duration of stable states. The typical timeout duration in transitional states is in the order of seconds.

TABLE I. IMPORTANT STATES OF THE SESSION LIFE-CYCLE

State	Membership of *ep* in relations Conn and Nego	Type of the state
Idle state	*ep* is present neither in Nego nor in Conn	stable state
Session negotiation	*ep* is participating in Nego relation	transitional state
Session establishment	*ep* is participating in Nego relation	transitional state
Active state	*ep* is participating in Conn relation	stable state
Session modification	*ep* is participating in both Nego and Conn relations	transitional state
Session teardown	*ep* is participating in Conn relation	transitional state

The session life-cycle is presented in Fig.1. The figure presents the case of a successfully established call, as in a general case, the session can be terminated in any state (i.e. there can be failures in session negotiation, establishment or modification). It can be seen that the important operations that change the state of the session are:

- negotiation
- establishment
- teardown

Each of three listed operations is realized through transmission of messages between participating nodes. Messages are complex data structures and will not be further analyzed here.

The telephony session protocol defines the state space: the set of states and the set of possible graphs of state transitions. Each realized session can be represented as a graph in the state space defined by the protocol. This graph depends also on the configuration of participating nodes and end user decisions.

There is a problem when representing telephony network mathematically. This problem lies in the time dimension of events in the network. Each established connection is present in the network through resources that are allocated in nodes participating in the session. Session establishment, modification or teardown does not happen simultaneously in all participating nodes - because there is a finite time interval required for transferring messages between participating nodes. Thus, there are moments when the session is in different states at different nodes participating in the same session. This situation occurs during all operations. The following list is not comprehensive since for each operation, we list one example:

- session establishment - one node is already in the session active state (stable state) while the other is still in the session establishing state (transitional state).
- session modification - the same as above
- session termination - one node is already in the idle state, while the other is still in terminating state.

We can conclude that session is viewed from two viewpoints - two sides participating in the session. In the general case, the session state viewed from the two viewpoints is not the same.

A potentially viable approach is to consider composite state which takes into account the states of all viewpoints. For example, we can say that $(x_1, ..., x_n) \in$Conn, when all *ep* nodes in the n-tuple consider the session to be in active state. We can also say that $(x_1, ..., x_n) \in$Nego, when at least one of $x \in \{x_1, ..., x_n\}$ considers ongoing session to be in the negotiation or modification state. Thus, we say that $(x_1,...,x_n) \in$Conn $\cap$ Nego, when at least one of $x \in \{x_1,...,x_n\}$ considers ongoing session to be in the modification state while all x consider session to be in the active state.

V. CONCLUSION

The relations given here model some important static and dynamic aspects of the telephony network. The network structure is a crucial static aspect and telephony sessions are the most important dynamic aspect.

We have identified the following three important relations (physically connectable - PhyC, connected - Conn, and negotiating - Nego) defined over the set of network nodes. The call can be interpreted as an equivalence class of relation Conn. Some aspects of the session life-cycle are expressed in terms of two relations (Conn, Nego), as well as the nature of states in the life-cycle.

The relations given here are far from the comprehensive model. However, they can be a foundation of a more comprehensive model that would include more sophisticated features and that will make the model usable for practitioners. Those features are related to the quality of service (QoS) and security (privacy protection, authentication, denial of service (DoS) prevention and/or mitigation) – and are directions for future work.

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