

# VOICE: A Decentralized System for Ubiquitous VoIP Services

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**Abstract.** We propose VOICE (Wireless vOICE), a fully distributed architecture capable of supporting and offering ubiquitous voice services over wireless IP-based data networks. Existing WiFi infrastructure - deployed by enterprises or even individuals - will be supplemented with software modules based on open standards in order to support VoIP. Furthermore, a new breed of VoIP providers by aggregating these currently isolated access networks will be able to offer alternative and advanced services to reachable end-users. These services would in certain cases completely bypass traditional telephony and cellular providers and in others smoothly cooperate with them. We will describe the overall architecture and the necessary modules that should be in place for realizing a system that creates a marketplace for communication services.

## 1 Introduction

VoIP (or IP Telephony) refers to the usage of IP technology for realizing telephone calls, as well as other more advanced communication services. It is widely believed that it has a major role in the evolution of the telecommunications industry from a centralized terrain to a competitive value chain of highly specialized providers. Voice nowadays is considered 'just another application', not a service totally controlled by the user's provider. This setting implies increased control to end-users, as they can deploy their own telephony servers and use their Internet connectivity to route calls to non-local destinations.

We believe the main technical reasons for this change in telecommunications market is a) the decoupling of end users from a single **identifier** and b) the ability of recent devices to use multiple access networks. These two developments acted as an antidote for monopoly termination; a situation that can arise when only a single provider can terminate calls to that user.

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Traditionally the user identifier was a phone number that was tightly related to user's location in the provider's network (like the local telephone switch of PSTN). Today, SIP and H.323 addresses can be used for identifying the destination party, and as a mean to resolve the ultimate network address (**locator**) where packets will be directed to. But, a mapping system must be in place in order for a provider to **search for alternative identifiers** and select the most appropriate (if more than one exist). Today a number of such systems are operational, including (Public) ENUM, Carrier ENUM and .tel domain.

The next step after collecting one or more identifiers for the call destination is to **search for network locators** (usually an IP address) of the subsequent signaling node where the call will be routed. There are cases when the next node (another signaling server, a telephony gateway or the destination) is not unique. This may be the case of users with multi-mode end devices, for example a 3G mobile phone with WLAN interface. Furthermore, imagine an end user that has activated multiple devices at the same time (i.e. home and office) and thus the incoming call notification must be forked to all available network addresses. Perhaps the more usual case is when the destination is a PSTN subscriber and there are multiple gateway providers that can terminate the call, each one with different cost and quality attributes (i.e. call blocking probability). Access to this information is achieved by specific routing protocols available today, like TRIP, H.501 and DUNDi. We propose a system, called VOICE, that has the potential to give providers another call routing alternative. Within the proposed network, currently isolated existing WLANs (e.g. hotspots deployed by cafes and restaurants) will form the access network upon which new attractive services will emerge and offered by interested parties. WiFi providers will be able to exploit their infrastructure in a flexible way by generating new business streams, while ITSPs (Internet Telephony Service Providers) will have the chance to provide innovative services to their customers and explore new business possibilities. We are interested mainly in Wireless LANs due to their widespread adoption by dedicated providers, municipalities and even individuals.

Calls from and to users within its coverage will be efficiently routed and in certain cases it will be possible even to completely bypass traditional fixed telephony and cellular network providers. Note that VOICE provides a mapping mechanism that allows incoming VoIP calls to end users, not only outgoing calls that is the usual case with traditional VoWLAN (Voice over WLAN) services. But, since it is unrealistic to assume that the proposed network will be sufficient enough to route every call by using exclusively its own pool of resources, trustworthy interfaces with traditional providers will be negotiated and put in place, once the network has recognized size and value.

The proposed system is characterised by its open and dynamic nature, since it facilitates interested parties to join and leave it by following flexible although well-defined procedures. On top of that, a solid management structure that will set the necessary rules for ensuring the operational integrity of the entire network will be established, covering a wide range of topics such as authentication of users, pricing, contract negotiation, accreditation of participants etc.

Flexible contract negotiation can have significant impact in VoIP value chain. Providers want to be compensated for their participation in call setup, either by direct payments or through a reciprocative mechanism. But this requires an established arrangement before hand and although VoIP peering is desirable for providers of the same size, currently it is very limited due to the complexity and time consuming process of negotiating interconnection terms. So they currently find it beneficial to enter in an agreement with wholesale ITSPs, also known as IXCs. This resulted in most ITSPs rejecting to process direct incoming calls from non-trusted providers.

The core functionality of WOICE is actually a mechanism for mapping between user identifiers like the different flavours of ENUM do. Actually it has many similarities with Carrier ENUM systems where routing information is shared between interconnected ITSPs, but in our case end users have given their consensus. However the dynamic nature of the routing information (mobile phone number to a SIP address for example) requires an automated procedure for finding if such mapping exists, either on demand or by protocols like TRIP and H.501.

We should note that WOICE is along the same line as other collaborative VoIP systems that are becoming widely popular (Skype, FWDout). Furthermore, Anagnostakis [3] has proposed a reciprocative mechanism for dealing with the 'free-riding' issue between VoIP peers that terminate calls to the PSTN. Similarly, Efstathiou and Polyzos [1], deal with the same issue in the Wireless LAN roaming context by using a token-exchange based accounting mechanism that incentivizes participating providers to offer network access to each other's customers.

This paper is organized as follows; Section 2 describes the general architecture and gives a simple usage scenario. The functionality of WOICE Aggregators and the 'middleware' functionality that is distributed in the WOICE network is summarized in Section 3 and 4 respectively. Finally, we conclude in Section 5.

## 2 The WOICE Architecture

We propose a two-layer, hierarchical system where Hotspot owners (WOICE WISPs) and WOICE ITSPs form the lower and upper layer respectively. From a functional point of view, the basic entity of the proposed system is the WOICE Aggregator. That is, an ITSP that can form her artificially owned access network by setting up bilateral agreements with WOICE WISPs and thus being able to terminate calls with low cost from/to the mobile telephones of users that enter one of the associated hotspots. In order to increase the value offered to its customers she can also set up agreements with traditional telecommunication providers (PSTN, cellular, ITSPs) in order to accept or place outgoing calls.

There exist two main options for WOICE Aggregators. In the first case, they could choose to act independently trying to attract the maximum possible number of hotspot owners. The other option is the creation of a collaborative VoIP network, a peer-to-peer community, of 'small' WOICE Aggregators which will

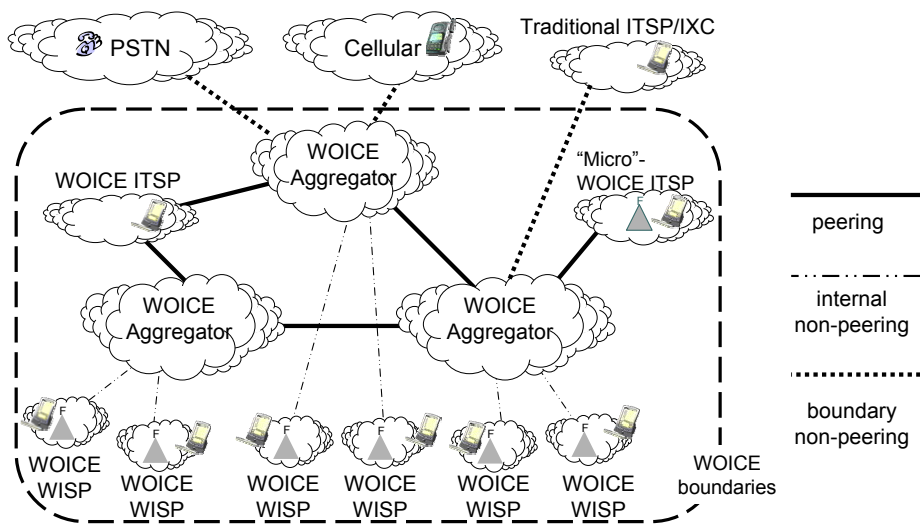


Fig. 1. The General VOICE architecture

exchange internally the call routing information of their networks in order to present aggregated call termination tables to external entities and be able to make profitable agreements with them. Note that in this case there is no restriction on the size of the enterprises that could participate in this VOICE network; even single VOICE WISPs with VoIP capabilities could play this role. Of course, besides the above extreme options there are many intermediate situations which one could think, such as a hierarchy of VOICE aggregators where some of them act as super-peers in the VOICE community.

Figure 1 shows a possible scenario of a VOICE network and its business relationships with traditional providers. The participants are of different types and size, showing the systems ability to support a large variety of business models. Each VOICE Aggregator has partnerships with a number of VOICE WISPs and may be interconnected with traditional providers. Other members of this VOICE community include a VOICE ITSP and a 'micro' VOICE ITSP. The former is a traditional ITSP with existing customer base and partnerships while the latter is a WISP with VoIP capabilities.

Figure 2 shows a usage scenario of VOICE. Lets assume that users A, B have dual-mode mobile phones that support the SIP signaling protocol. When, for example, 'User A' enters HotSpot, the VOICE WISP collects and verifies her mobile number. At the next step, the VOICE Aggregator is being informed about the new reachable number and subsequently advertises reachability to the VOICE ITSP so that calls can take place.

When 'User B' calls 'User A' on her traditional mobile phone number, the VOICE ITSP of 'User B' will search her routing table and find that the call can be terminated using end-to-end VoIP through VOICE Aggregator. VOICE

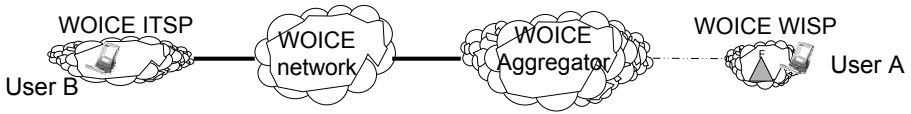


Fig. 2. A call placement scenario through WOICE system

ITSP sends a call setup notification (for example using SIP) to the WOICE Aggregator and exchange accounting information. Finally, WOICE ITSP and WOICE Aggregator rate each other and this reputation information is stored inside the WOICE network in a distributed way for enforcing rules in the future.

### 3 The WOICE Aggregator

We will present the functionality of WOICE incrementally. We will first describe the functionality for an independent WOICE Aggregator (no peer partners) relative to the interfaces that should be supported with his WOICE WISPs. Then we will analyse how this functionality may be implemented in a distributed fashion, so as to enable the creation of a large WOICE network and thus a variety of business models particularly attractive to a wide range of participants.

Figure 3 shows a high-level view of the core WOICE software modules and their interfaces that allow the interoperation of a standalone WOICE Aggregator with his partners. This resembles the case of a WOICE network with one big WOICE Aggregator.

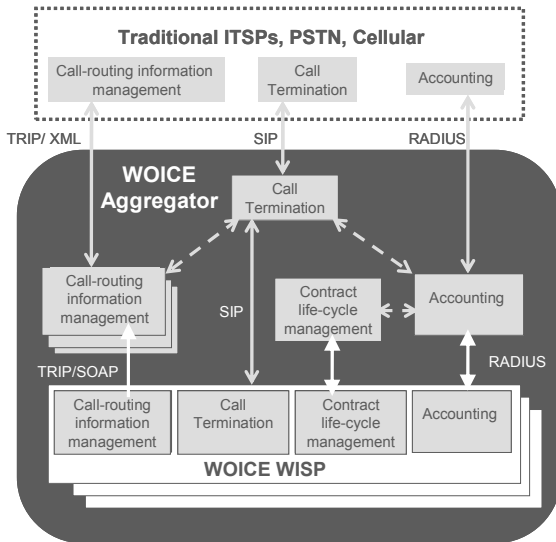


Fig. 3. Non-peer Interoperation

### 3.1 Contract Life-Cycle Management Module

VOICE Aggregators may have many bilateral contracts with VOICE WISPs. The most significant part of these agreements is usually the general contract terms, as well as the pricing scheme being used and each party's obligations (SLAs). For example, the two parties could agree that a Hot Spot visitor can place VoIP calls for free from his WiFi-mobile phone and the venue owner will be charged. In parallel, the same visitor can receive VoIP calls for free and the venue owner will receive a payment for using his Internet connection. This module automates the negotiation procedure by offering templates and guiding the transition from one contract term to another, until the agreement is either established or abandoned. The messages exchanged between the two providers represent their initial requests or bargaining responses.

### 3.2 Call-Routing Information Management Module

During the valid period of an agreement and every time a mobile-user enters the Hot Spot of a VOICE WISP the new routing information is propagated throughout the system and the external partnered entities. These actions are performed by the interoperation of each partner's Call-Routing Information Management module. Due to the importance of the VOICE Aggregator for the system's operation his module has interfaces for both intra-domain and inter-domain communication. The messages sent from each VOICE WISP to the VOICE Aggregator can be performed with TRIP or by web services technology (SOAP messages). The messages sent from the VOICE Aggregator to external partners can be performed with a call-routing protocol like TRIP or by XML files. In this way, the VOICE Aggregator can terminate calls from traditional providers to mobile users that reside in associated Hot Spots.

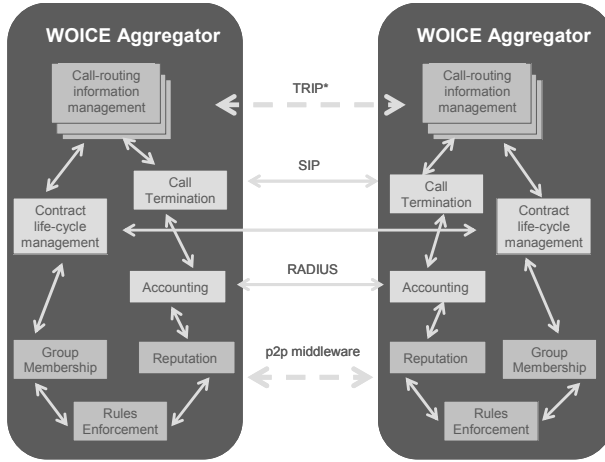
In [2] it has been concluded that in many cases providers can have the ability and incentive to ask for more than one providers to try and connect the same call (a technique called forking). This gives an incentive to providers to route calls to low-cost systems even when the call blocking probability is very high. Thus, even if providers don't have access to this routing information, they could still attempt to route the call through VOICE and a traditional provider.

### 3.3 Call Termination Module

Every time a VOICE Aggregator participates in a VoIP call (either initiated by or terminated to a mobile-user) the Call termination module of a number of entities must interoperate. These modules can reside on the users mobile phone, the VOICE Aggregators / external ITSPs signalling server or on a PSTN/cellular telephony gateway. The messages exchanged refer to a signalling protocol and perform call management (establishment, modification and termination).

### 3.4 Accounting Module

During the valid period of a bilateral agreement and every time a partner requests a service that is supported by the contracts terms of service, the two parties must



**Fig. 4.** Peer Interoperation

keep information regarding this transaction. For example, a WOICE WISPs request to WOICE Aggregator for terminating a mobile users call would drive the two partners to track the acceptance (or not) of request, duration, charges (or revenue shares) among others. This task is performed by the Accounting module of each provider that is based on the RADIUS accounting protocol for example.

## 4 The WOICE Network

We would like to enable the creation of a decentralized network of WOICE aggregators. For this to happen, one needs to implement specific 'middleware' modules that would support a) the distributed, efficient and accurate call routing information management, and b) group management, decision making, and enforcement of specific community rules that are needed to discourage selfish behaviour and ensure certain levels of QoS. In this way, the formation of a community is more likely to succeed. In the following, we summarize the functionality of each module.

### 4.1 Group Membership and Search

Group membership includes the fundamental procedures of joining and leaving the community, the management of identities and certificates, and the ability to exclude members. Furthermore, we will support the ability to search for WOICE aggregators with specific characteristics for the establishment of agreements. Examples of such attributes are reputation score and location.

## 4.2 Reputation

Reputation can be considered as a distributed accounting mechanism since its value for a specific peer expresses his 'aggregated' behaviour in the past. Reputation values are constructed through ratings of individual peers after a completion of a transaction or through monitoring of the behaviour of certain peers in the network. Issues such as 'cheap pseudonym' could be mitigated due to the ability to recognize domain names.

## 4.3 Rules Configuration and Enforcement

The existence of community rules which will ensure the desired behaviour of the participants will play a major role to the success of VOICE. However, we would like to offer substantial freedom to communities to configure these rules according to their needs (e.g. through voting or off-line configuration by the system designer, or dynamic adaptation according to some predefined measures). Examples of such parameters of community rules could be the constraints on the reputation threshold for a peer to be able to act as a super peer and the pricing policy of the community.

## 5 Conclusions

We described a collaborative system that allows offering of ubiquitous VoIP services to end users and flexible business models for its participants. In order for these targets to be met a) routing information is exchanged between providers that maps traditional user identifiers to VoIP identifiers and b) supplementary functionality is introduced that aligns participants' incentives.

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