

# Design and Implementation of "Handipum" WebRTC-based Conferencing Service in ASNET-AM Network

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## Abstract

This paper presents the results of research and deployment experience with design and implementation of the ASNET-AM WebRTC-based audio/video conferencing service, called "Handipum" (handipum.asnet.am). Since the WebRTC-services become widely used, this service can provide a modern browser-based conferencing environment for ASNET-AM users.

**Keywords:** Real-Time Communications, RTC, WebRTC, Selective Forwarding Unit, SFU, Conferencing, Multimedia, Middleware

## 1. Introduction

WebRTC is a modern open source technology that enables web browsers with Real-Time Communications (RTC) capabilities via JavaScript APIs. It has been initially developed as a peer-to-peer technology, where clients (web browsers) can directly communicate without any intermediate infrastructure. This model is enough for creating basic applications but features such as group communications, media stream recording, media broadcasting or media transcoding are difficult to implement on top of it. For this reason, many applications require using some kind of intermediate media server. Conceptually, the WebRTC media server is just a kind of "multimedia middleware" (it is in the middle of the communicating peers) where media traffic passes through when moving from source to destinations. Media servers are capable of processing media streams and offering different types including group communications (distributing the media stream that one peer generates among several receivers), mixing (transforming several incoming streams into one single composite stream), transcoding (adapting codecs and formats between incompatible clients), recording (storing in a persistent way the media exchanged among peers), etc.

## 2. Architecture Models

There are in general 3 main architecture models of deploying a multiparty WebRTC-based video conferences: Mesh, Multipoint Conferencing Unit (MCU) & Selective Forwarding Unit (SFU). In Mesh multipoint architecture every participant sends and receives his media to all other participants. This variant is a very common technique used in WebRTC to build multipoint conferences. Advantages of mesh architecture are that it's simple to implement in WebRTC and requires very little backend infrastructure, keeping the resulting service cheap to operate. But it requires a lot of uplink bandwidth from each participant and thus can't scale to a large number of participants.

An MCU offers the ability to connect multiple participants in a single voice or video session. In MCU participants "speak" to a central entity, which mixes all inputs and sends out a single stream towards each participant. MCUs generally implement a mixing architecture and are expensive due to their need for a lot of processing power per session.

An SFU is capable of receiving multiple media streams and then decide which of these media streams should be sent to which participants. In SFU the participant sends his media to a central entity, which routes all incoming media as it sees fit to participants - each of them receiving usually more than a single stream.

So SFU can be treated as a lightweight WebRTC implementation, allowing for multiuser video communication by simply relaying the received multimedia flows to all call participants, without mixing into a composite stream. One of the effective open-source SFU solutions is Jitsi Videobridge. Together with other components from Jitsi project [1] it proved to be an interesting solution that was used as a basis for creating a WebRTC-based audio/video conferencing service in ASNET-AM network, called "Handipum" [2].

## 3. "Handipum" Conferencing Service Infrastructure

There are lots of websites offering online video conferencing solutions on the Internet today. Many have a user friendly interface and very good video and audio quality. Some even are free for use after registration. However, the downside of such solutions is that the data moves through a foreign server. Plus no Armenian interface is available anywhere in conferencing solutions.

Thus, it was decided to design and implement an own conferencing solution for ASNET-AM users, based on freely available solutions. After some comparative research, one of such solutions was found to be the Jitsi project, which provides an open source, encrypted, and sustainable interface. It works via web front end application called Jitsi Meet [3], thus making it possible to use any WebRTC-compatible web browser as a client. Such a modern flexible browser-based solution eliminates the need for a specific client and at the same time provides rich conferencing capabilities.

Two main components of "Handipum" Conferencing Service are – Jitsi Videobridge and Jitsi Meet. They work together to transfer video and audio between the users. Videobridge acts as a web real-time communication WebRTC-compatible SFU and allows for multiuser communication.

Jitsi Meet is a front end to Videobridge and is implemented in JavaScript. Jitsi Meet can be configured to be hosted on some HTTP server and in our case the Nginx web server [4] was used for that. In addition some implementation of XMPP server is required on the server side. The Prosody [5] lightweight, open source Jabber/XMPP server is the basic solution used as an XMPP server.

### 4. "Handipum" Conferencing Service Interface

"Handipum" Conferencing Service currently offers video and audio conferences in web browsers with real-time text chat, Youtube video sharing. Other additional services that are under development include: presentation sharing, screen sharing, shared document editing, VoIP integration.

We have done the localization of Jitsi Meet web interface and "Handipum" Conferencing Service is now available in Armenian. We have also provided the Armenian translation to the public [6]. So Jitsi Meet is now translated into 16 languages, among which Armenian is proudly presented, thanks to the ASNET-AM Team efforts.

Users can take part in meetings with WebRTC-enabled browsers. The current versions of Opera, Google Chrome, and Mozilla Firefox already support standard; additional functions such as desktop sharing require browser add-ons. Microsoft's Internet Explorer/Edge and Apple's Safari browsers do not have proper WebRTC support yet.

All conferencing communication is secure enough thanks to the HTTPS encrypted connections. Strong SSL Certificate with 4096-bit keys SHA-256 signature is used in "Handipum" Conferencing Service. Additionally each conference room created, can be locked by a password, to ensure limited access.

### 5. Conference Room Creation

Users can open a new room in "Handipum" Conferencing Service by opening its start page <https://handipum.asnet.am> (Figure 1) and typing a unique name for conference room. The room name should not contain spaces and should consist of only Latin letters and numbers.

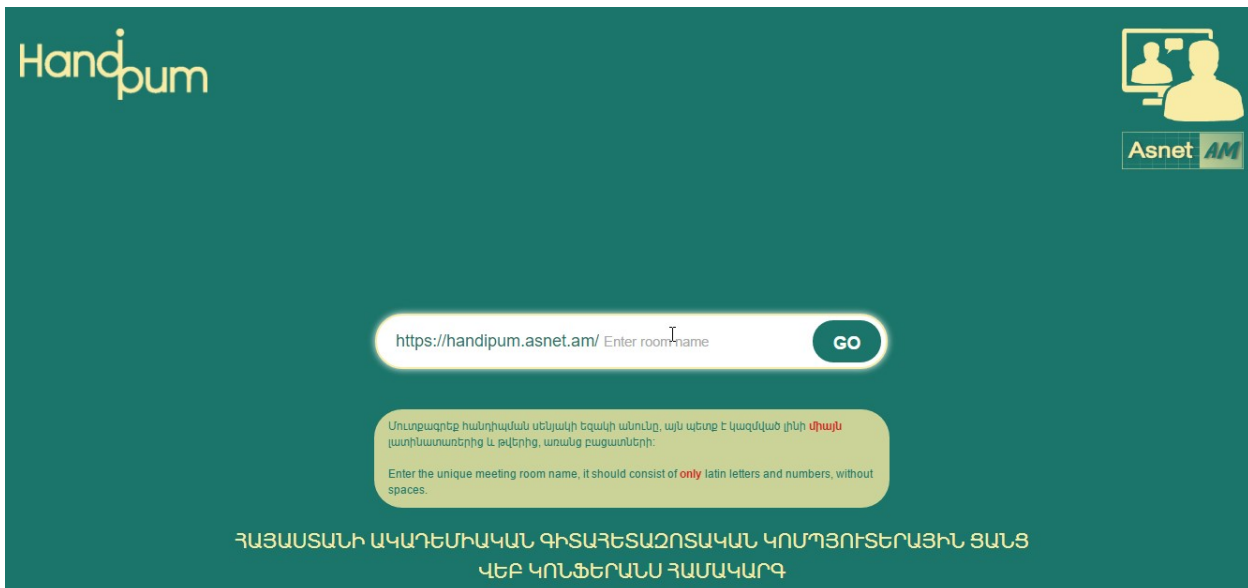


Fig. 1. "Handipum" Conferencing Service start page

After clicking the “GO” button the system creates a unique room interface as shown on Figure 2.

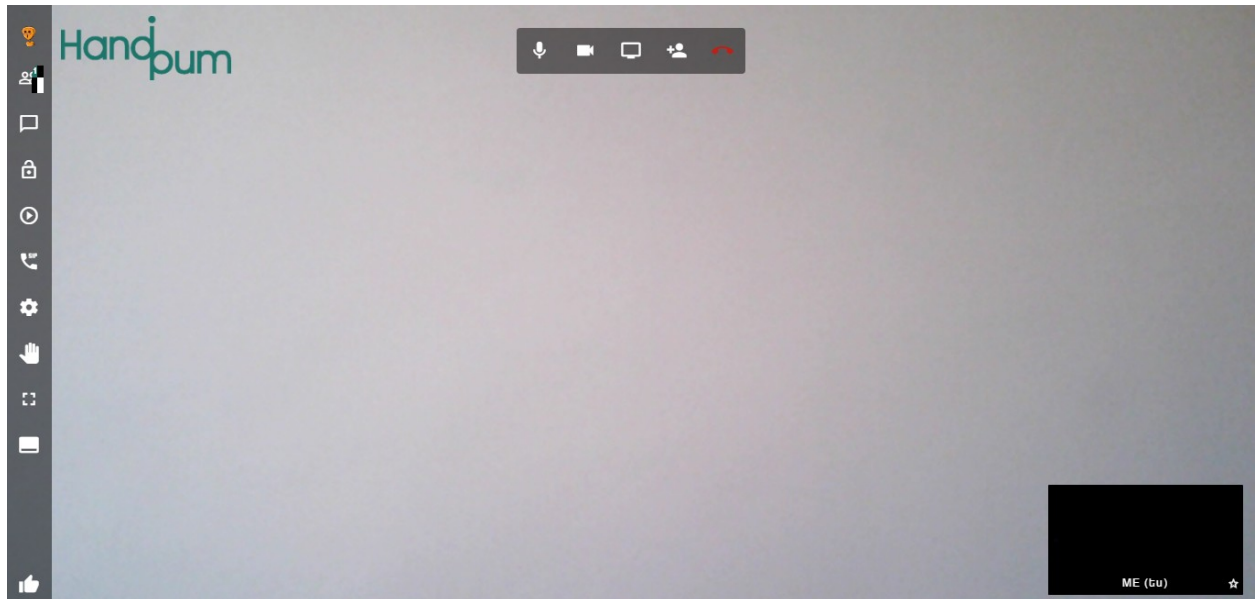


Fig. 2. "Handipum" Conferencing Service room interface

While entering the conference room, participants should confirm that they allow the "Handipum" Conferencing Service to access their camera and microphone. The initiator of the conference (the user, who initially created a new non-existing room) automatically receives moderator rights, can then protect the room with a password and mute or remove participants. Currently it is not possible to provide moderator rights to any other participant. All current room participants are displayed in separate mini-windows at the bottom, and the moderator is marked with an asterisk.

## 6. SIP Integration

"Handipum" service is configured to make instant calls from a conference room to some SIP number within ASNET-AM Internal IP Telephony service. SIP integration is implemented with Jigasi (Jitsi GAteway to SIP) [7]. Jigasi is a server-side application that allows regular SIP clients to join the "Handipum" conference.

Integration is made by setting a specific ASNET-AM SIP server account in the Jigasi configuration. A small phone icon appears in conference rooms in the web interface. Any participant can call someone at a SIP number and add him to current room conversation. Clicking on phone icon opens a box in which the inviting participant needs to enter the SIP number to call (Figure 3). If the individual being called picks up, he/she can take part in the audio conversation within that room. By this, users with SIP phones can take part in "Handipum" conferences.



Fig. 3. Inviting participant needs to enter the SIP number to call

## 7. Conclusion

Conferencing solution presented above is a modern WebRTC-based browser-driven system. It is very efficient and scalable. It is able to run thousands of simultaneous video streams at small bandwidth and CPU usage [8]. Before "Handipum" many of our users - scientists, scientific and technical associates, postgraduates and students used other tools such as Microsoft Skype or Google Hangouts for conferencing. Since the "Handipum" conferencing service is now open to our community and since it works directly from web browser and does not require any specific client installation, we hope it to become a viable alternative that perfectly matches our users' needs. ASNET-AM members were informed about the "Handipum" conferencing service available, by publishing the appropriate news at ASNET-AM website.

## References

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## WebRTC միջավայրում "Handipum" վեբ կոնֆերանս համակարգի նախագծում և իրականացում ASNET-AM ցանցում

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### Ամփոփում

Հոդվածում նկարագրված է նոր սերնդի վեբ կոնֆերանս համակարգի նախագծումն ու իրականացումը ASNET-AM ցանցում, որն իրականացված է WebRTC միջավայրում: WebRTC համակարգերը լայն տարածում են ստացել այն պատճառով, որ օգտագործողի համակարգչի վրա չեն պահանջում հատուկ ծրագրերի տեղադրում և

աշխատում են անմիջապես վեբ բրաուզերներում: ASNET-AM ցանցում ներկայումս գործող այդպիսի համակարգը, որի մասին խոսվում է հոդվածում, հիմնված է Jitsi բաց կոդով ծրագրային փաթեթի վրա: Նկարագրված են համակարգի հիմնական հնարավորությունները: Վեբ կոնֆերանս համակարգը անվանվել է "Handipum" և հասանելի է <https://handipum.asnet.am> հասցեով: Մինչև "Handipum" համակարգի ներդրումը, մեր օգտագործողներից շատերը, որոնց թվում են՝ գիտնականները, գիտական և տեխնիկական աշխատողները, ասպիրանտները, ուսանողները և այլ մասնագետները, օգտվում էին տարբեր այլ արտաքին կոնֆերանս համակարգերից: Այժմ, երբ "Handipum" համակարգը հասանելի է ողջ հայկական գիտակրթական հանրության համար և աշխատում է անմիջապես բրաուզերից, հույս ունենք, որ այն արժանի տեղական այլընտրանք կլինի մեր օգտագործողների համար:

## **Разработка и реализация сервиса конференций "Handipum" на основе протоколов WebRTC в сети ASNET-AM**

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### **Аннотация**

В статье описана разработка и реализация нового поколения сервиса веб конференций в сети ASNET-AM на основе протоколов WebRTC. Реализация сервисов веб конференций в среде WebRTC в настоящее время пользуется большой популярностью по причине отсутствия необходимости установки специального клиентского приложения, в качестве которого выступает обычный веб браузер. Действующая в настоящее время в сети ASNET-AM услуга веб конференций на основе протоколов WebRTC базируется на активно развивающемся проекте с открытым кодом Jitsi. Описаны основные возможности сервиса. Сервис назван "Handipum" и функционирует по адресу <https://handipum.asnet.am>. До внедрения сервиса "Handipum", многие из наших пользователей - ученые, аспиранты и студенты, пользовались прочими внешними возможностями для конференций. Поскольку сервис "Handipum" в настоящее время открыт для армянской научно-образовательной среды и работает прямо из браузера, мы надеемся, что он станет достойной альтернативой для наших пользователей.