CROATIA

NG112 Project

Report

In this Final Report for EENA's Next Generation 112 Project, discover the findings of the consortium based in Croatia.

CROATIA

NG112 Project Report

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EXECUTIVE SUMMARY

The Next Generation 112 (NG112) architecture has the potential to transform emergency response by enabling Public Safety Answering Points (PSAPs) to receive a wide range of data. Emergency services would not only have access to voice, but also location information, real-time text, video calls and other data. Not only would this solve many problems faced by call takers, such as locating people in distress, but it would also help improve accessibility of emergency services for people with disabilities.

The NG112 project, launched by EENA, aimed to demonstrate the value of the NG112 architecture in real-life environments. Three consortia were selected, involving five different countries. Ultimately, the project aimed to help to launch the deployment of NG112 and improve emergency response for citizens.

This report summarises the testing activities and lessons learnt of the Croatian consortium. The Croatian consortium tested three use cases in the Internet Protocol (IP) environment: emergency voice call with location, emergency video call, and text chat. Various challenges were overcome during the implementation of the project, resulting in numerous recommendations and considerations which may assist other countries in the deployment of the NG112 architecture. The value of additional data and improved accessibility were successfully demonstrated by the Croatia Consortium.

EENA launched the NG112 Project to test and deploy the technical architecture enabling NG112 in different European countries, with a focus on demonstrating its use in real-life environments.

Three consortia were selected for the project:

CELESTE : Austria, Italy, Denmark

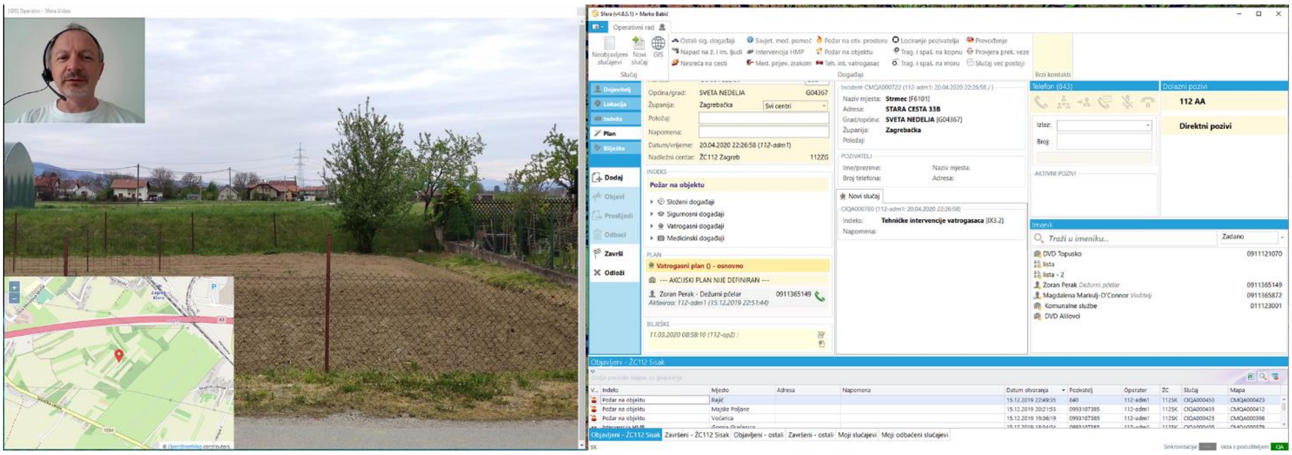
Croatia

Turkey

1 | OVERVIEW

The Croatian consortium – comprised of the Ministry of the Interior and KING ICT – tested 3 use cases as part of the NG112 project.The testing was carried out in two regions of Croatia - Zagreb county and the City of Zagreb – on the Sfera CAD system, which is implemented in three quarters of 112 PSAPs in Croatia. The test cases were:

- Emergency voice call with location sent over IP
- Emergency video (and voice) call with location over IP
- Text emergency call with location over IP



Općina/grad: SVETA NEDELJA G04367
Županija: Zagrebačka Svi centri
Položaj:
Napomena:
Datum/vrijeme: 20.04.2020 22:26:58 (112-adm1)
Nadležni centar: ŽC112 Zagreb 112ZG

INDEKS
Požar na objektu
- Složeni događaji
- Sigurnosni događaji
- Vatrogasni događaji
- Medicinski događaji

PLAN
Vatrogasni plan () - osnovno
--- AKCIJSKI PLAN NIJE DEFINIRAN ---
Zoran Perak - Dežurni pčelar 0911365149
Aktivirao: 112-adm1 (15.12.2019 22:51:44)

BILJEŠKE
11.03.2020 08:58:10 (112-op2) :

Incident-CMQA000722 (112-adm1: 20.04.2020 22:26:58 /)
Naziv mjesta: Strmec [F6101]
Adresa: STARA CESTA 33B
Grad/općina: SVETA NEDELJA [G04367]
Županija: Zagrebačka
Položaj:

POZIVATELJI
Ime/prezime:
Broj telefona:
Naziv mjesta:
Adresa:

Novi slučaj
CIQA000760 (112-adm1: 20.04.2020 22:26:58)
Indeks: Tehničke intervencije vatrogasaca [IX3.2]
Napomena:

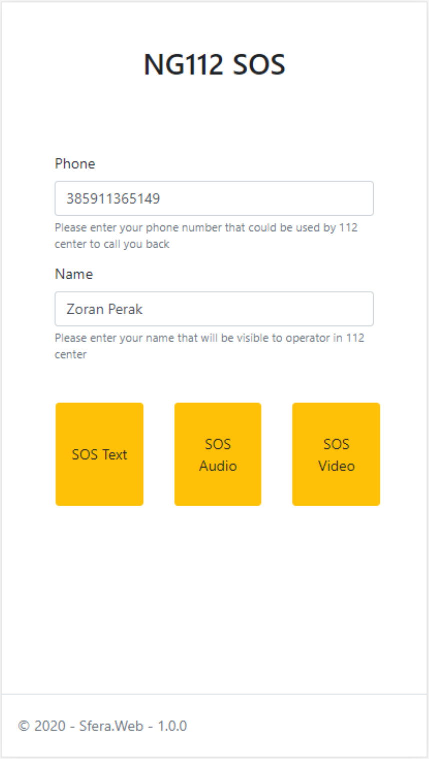
DVD Topusko	0911121070
lista	
lista - 2	
Zoran Perak *Dežurni pčelar*	0911365149
Magdalena Markulj-O'Connor *Voditelj*	0911365872
Komunalne službe	011123001
DVD Alilovci	

V...	Indeks	Mjesto	Adresa	Napomena	Datum otvaranja	Pozivatelj	Operater	ŽC	Slučaj	Mapa
	Požar na objektu	Rajić			15.12.2019 22:49:35	840	112-adm1	112SK	CIQA000450	CMQA000423
	Požar na objektu	Majske Poljane			15.12.2019 20:21:53	0993107385	112-adm1	112SK	CIQA000439	CMQA000412
	Požar na objektu	Vočarica			15.12.2019 19:36:19	0993107385	112-adm1	112SK	CIQA000425	CMQA000398

Figure 1: Sfera (PSAP) during video call. Screenshot from two monitors.

These use cases are described in detail, with a step-by-step process for each, in Chapter 3. In Chapter 4, we examine the NG112 architecture as applied to the Croatia test cases, including details of the basic flow of the call and the protocols used between the system components.

The emergency communications could be initiated through a webpage. The consortium identified that a key benefit of this is that the person in distress does not need to have pre-installed an application to initiate a communication, as it can be accessed via a standard web browser. The URL should be widely publicised.



NG112 SOS

Phone

385911365149

Please enter your phone number that could be used by 112 center to call you back

Name

Zoran Perak

Please enter your name that will be visible to operator in 112 center

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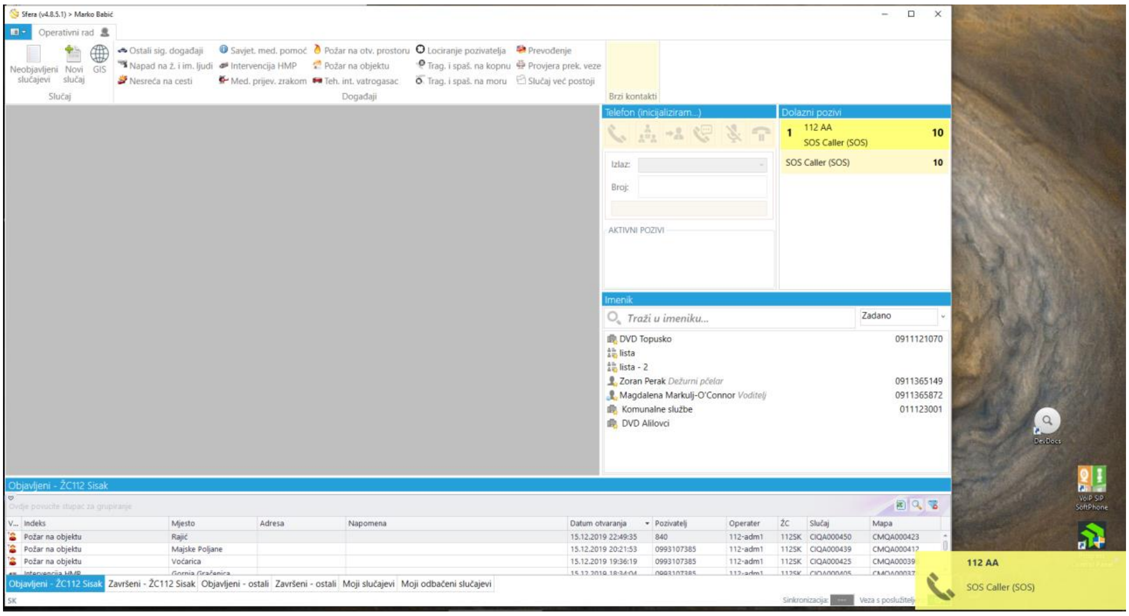
Figure 2: PWA initial screen

On the home screen, the citizen is asked to select the service that they require. If it is their first use of the service, they should also enter their phone number and name. Once the request has been established, the communication is placed in a waiting queue and the operator in the 112 PSAP is alerted. More information about the testing can be found in Chapter 5.



1	112 AA SOS Caller (SOS)	10
	SOS Caller (SOS)	10

DVD Topusko	0911121070
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lista - 2	
Zoran Perak *Dežurni pčelar*	0911365149
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V...	Indeks	Mjesto	Adresa	Napomena	Datum otvaranja	Pozivatelj	Operater	ŽC	Slučaj	Mapa
	Požar na objektu	Rajić			15.12.2019 22:49:35	840	112-adm1	112SK	CIQA000450	CMQA000423
	Požar na objektu	Majske Poljane			15.12.2019 20:21:53	0993107385	112-adm1	112SK	CIQA000439	CMQA000412
	Požar na objektu	Voćarica			15.12.2019 19:36:19	0993107385	112-adm1	112SK	CIQA000425	CMQA00039
	Intervencija HMP	Gornja Gračenica			15.12.2019 18:34:04	0993107385	112-adm1	112SK	CIQA000405	CMQA00037

112 AA
SOS Caller (SOS)

Figure 3: Sfera (PSAP system) receiving web call

The consortium concluded that video stream is of great value to emergency services for assessment of the situation and resource allocation. Some challenges were encountered regarding the implementation of the text solution, but this was overcome by using a WebRTC data channel.

The main considerations for implementation were identified as:

- Anonymous vs registered access to 112
- Speed of internet access
- Cost of internet access

These are explained in detail in Section 6.1.

The next steps for the Croatian consortium include piloting the project in production, first by establishing an ESInet in a single PSAP that would cover the whole country. In a second phase, other PSAPs would be gradually connected to the ESInet.

2 | CONSORTIUM

Ministry of the Interior - Croatian Civil Protection Directorate

The Ministry of the Interior of the Republic of Croatia is a central authority of the state administration responsible for the civil protection system. Within the Ministry, the Civil Protection Directorate is an administrative organisation which prepares, plans and manages the operational forces, coordinating the functioning of all the civil protection system. The Civil Protection Directorate is also responsible for management of 112 system.

KING ICT

KING ICT is the leading systems integrator in SEE (present in Croatia, Serbia, Bosnia and Herzegovina, Kosovo and Macedonia) and the leader in developing and implementing innovative enterprise solutions.

King ICT software solutions range from those used by a great number of people in everyday life to highly specialised ones, such as KING SFERA, e.Police, e.Goverment, SIS.

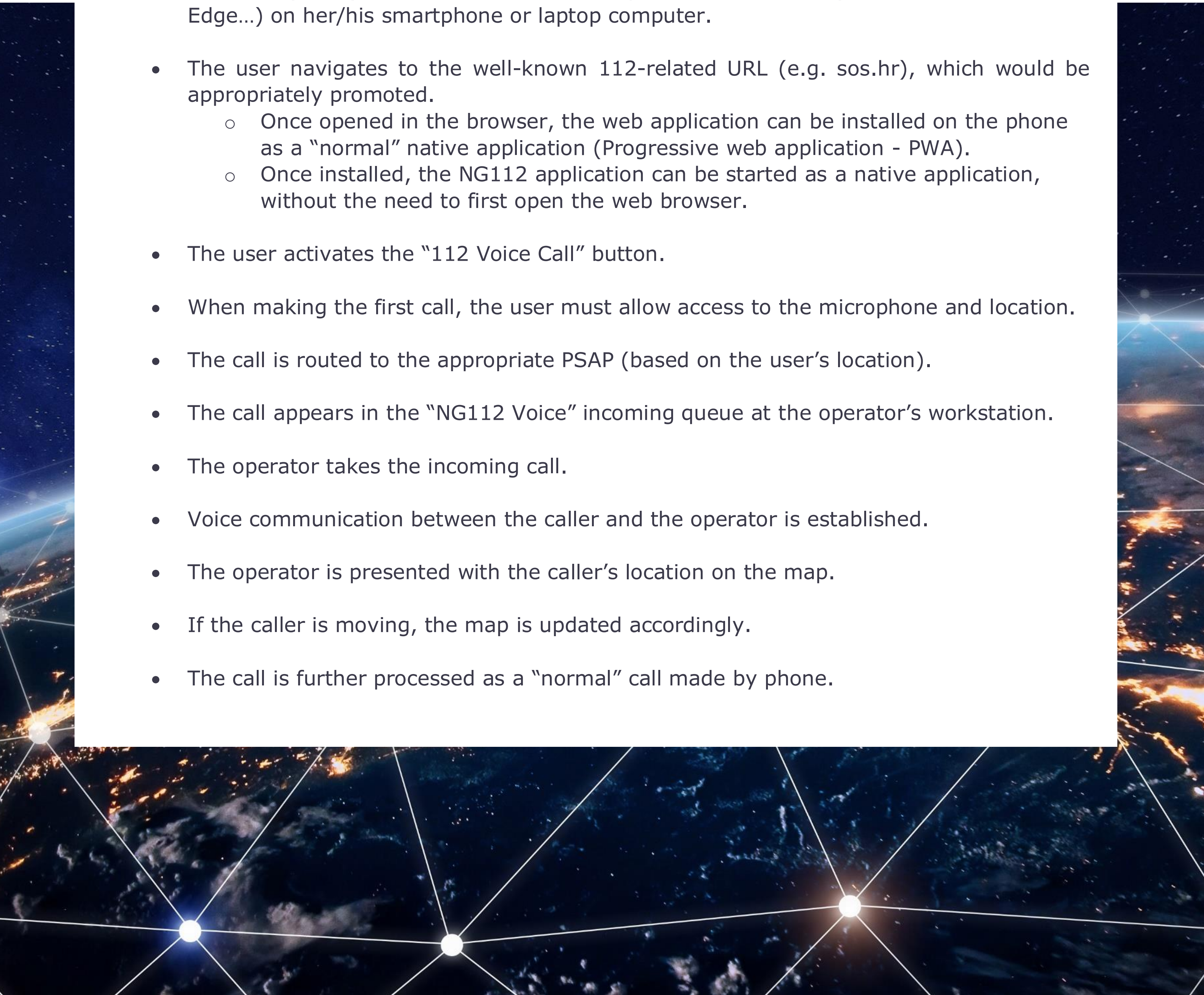
KING SFERA is a software solution that enables communication, coordination, and management in emergency situations.

3 | USE CASES

The main purpose of this project was to test the possibility of using already available standards based modern technologies to access 112 emergency services without the need for previous preparation in the form of installation of a specialised 112 mobile application, registration etc. The solution should enable voice, video and text communication together with real time location tracking available to PSAP operators.

1. EMERGENCY VOICE CALL WITH LOCATION SENT OVER IP, FROM SMARTPHONE OR LAPTOP COMPUTER

- The user (caller) starts the standards-compliant web browser (Chrome, Firefox, Safari, Edge…) on her/his smartphone or laptop computer.
- The user navigates to the well-known 112-related URL (e.g. sos.hr), which would be appropriately promoted.
  - Once opened in the browser, the web application can be installed on the phone as a "normal" native application (Progressive web application - PWA).
  - Once installed, the NG112 application can be started as a native application, without the need to first open the web browser.
- The user activates the "112 Voice Call" button.
- When making the first call, the user must allow access to the microphone and location.
- The call is routed to the appropriate PSAP (based on the user's location).
- The call appears in the "NG112 Voice" incoming queue at the operator's workstation.
- The operator takes the incoming call.
- Voice communication between the caller and the operator is established.
- The operator is presented with the caller's location on the map.
- If the caller is moving, the map is updated accordingly.
- The call is further processed as a "normal" call made by phone.



2. EMERGENCY VIDEO (AND VOICE) CALL WITH LOCATION OVER IP FROM SMARTPHONE OR LAPTOP COMPUTER

- The user (caller) starts the standards-compliant web browser (Chrome, Firefox, Safari, Edge…) on her/his smartphone or laptop computer.

- The user navigates to wthe ell-known 112 related URL (e.g. sos.hr), which would be appropriately promoted.
  - Once opened in browser, the web application can be installed on the phone as a “normal” native application (Progressive web application - PWA).
  - Once installed NG112 application can be started as native application, without need to first open web browser.

- The user activates the “112 Video Call” button.

- When making the first call, the user must allow access to the camera, microphone and location.

- The call is routed to the appropriate PSAP (based on the user’s location).

- The call appears in the “NG112 Video” incoming queue at the operator’s workstation.

- The operator takes the incoming call.

- Video (and voice) communication between the caller and the operator is established.
  - On the operator’s side, a new window is opened with bigger video from the caller and smaller video from the operator.
  - On the caller’s side, if it is a smartphone, the rear camera is turned on by default. The caller is presented with a bigger video from the rear camera and a smaller video from the operator. The caller can switch the camera between the front and rear. If it is a laptop, then the caller is presented with a bigger video of the operator and a smaller video of the caller from the front camera.

- The operator is presented with the caller’s location on map

- If the caller is moving, the map is updated accordingly.

3. TEXT EMERGENCY CALL WITH LOCATION OVER IP FROM SMARTPHONE OR LAPTOP COMPUTER

- The user (caller) starts the standards-compliant web browser (Chrome, Firefox, Safari, Edge…) on her/his smartphone or laptop computer.

- The user navigates to well-known the 112 related URL (e.g. sos.hr), which would be appropriately promoted.
  - Once opened in the browser, the web application can be installed on the phone as a "normal" native application (Progressive web application - PWA).
  - Once installed, the NG112 application can be started as a native application, without the need to first open the web browser.

- The user activates the "112 Text Call" button.

- When making the first call, the user must allow access to location.

- The call is routed to the appropriate PSAP (based on the user's location).

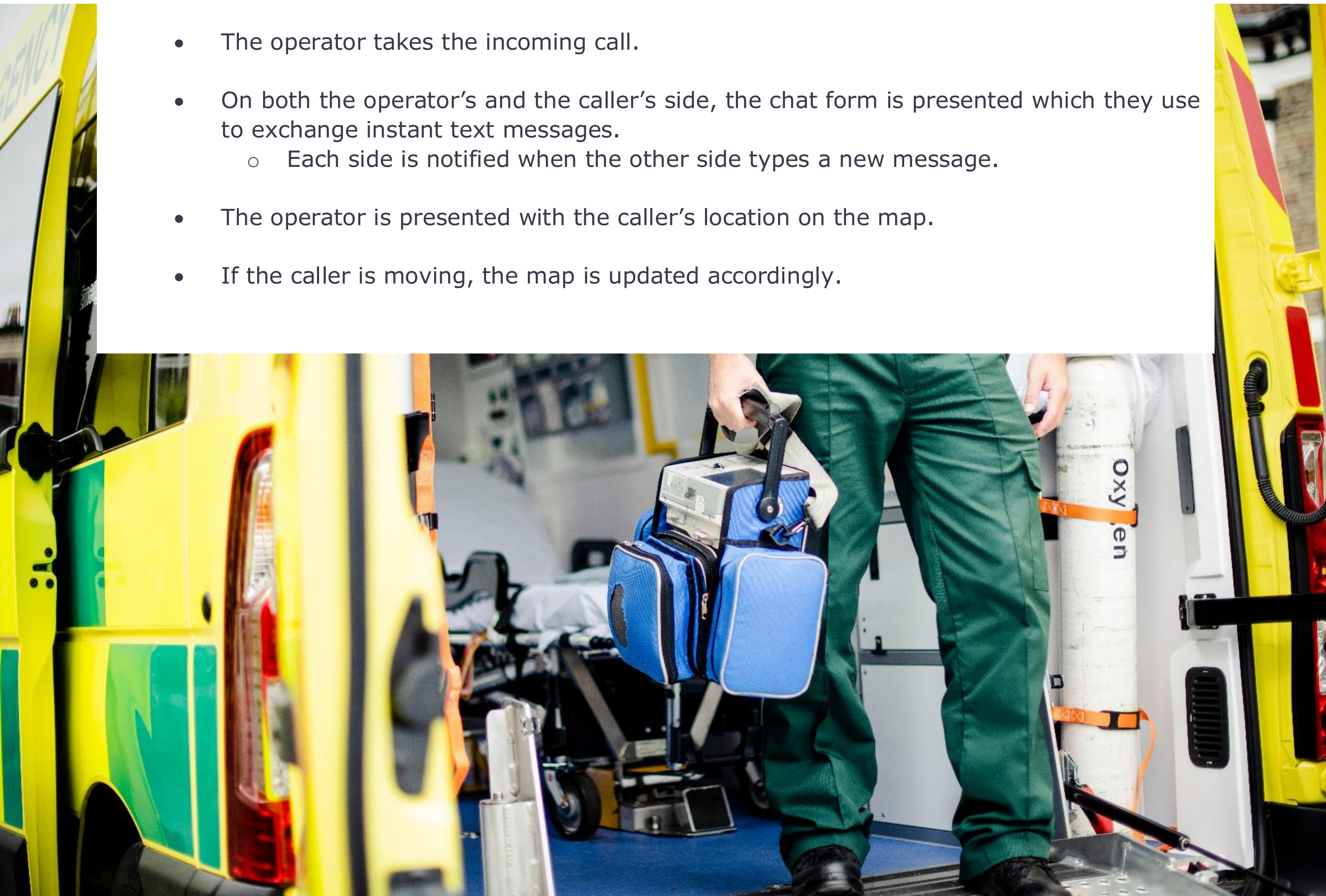
- The call appears in the "NG112 Text" incoming queue at the operator's workstation.

- The operator takes the incoming call.

- On both the operator's and the caller's side, the chat form is presented which they use to exchange instant text messages.
  - Each side is notified when the other side types a new message.

- The operator is presented with the caller's location on the map.

- If the caller is moving, the map is updated accordingly.



4 | ARCHITECTURE

The solution is based on standard technologies that are ubiquitous and already available on both Android and iOS mobile platforms, as well as on laptop and desktop computers. The key underlying technology is WebRTC.

Its architecture is also based on EENA's *Long Term Definition Standards document*[1].

Figure 4 shows an overall diagram of the solution. The caller uses a smartphone or laptop computer to access the SOS website with a standards-compliant browser (Chrome, Firefox, Safari, new Edge, Opera…). The SOS website provides a Progressive Web Application (PWA) that contains a SIP User Agent (UA), i.e. SIP phone implemented with JavaScript. In this case, we have used the jsSIP open source library for handling SIP and WebRTC.

In testing, setup used in this project BCF between ESInet and PSAP is not used.

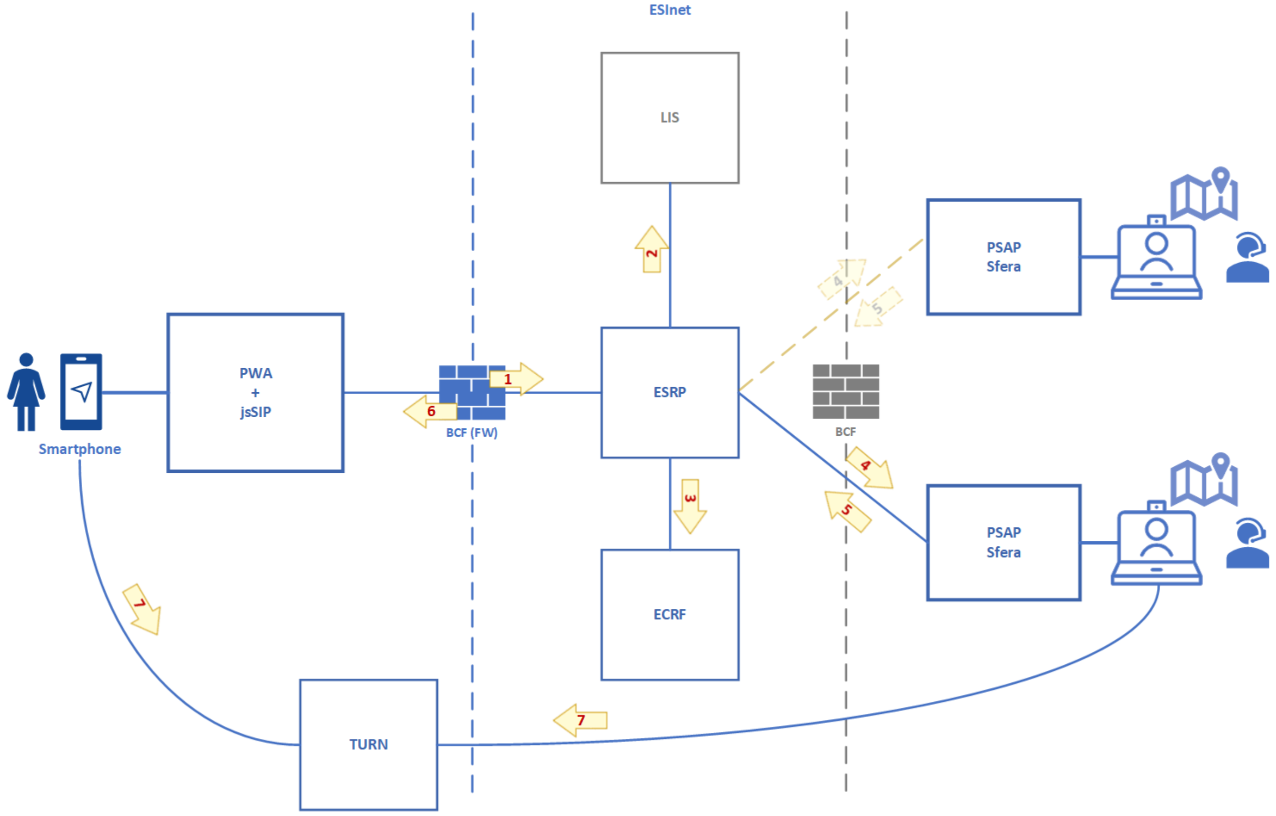


Figure 4: Solution architecture

[1] *https://eena.org/document/ng112-long-term-definition-standard-for-emergency-services/*

Here is the basic flow of the the call:

1. SIP client (jsSIP) sends INVITE to ESRP SIP Proxy server via WebSocket. Message contains SDP (WebRTC Offer) and location (PIDF-LO).

2. ESRP updates LIS server with new location (PUBLISH).

3. Get most appropriate PSAP based on caller location (LoST).

4. Proceed to most appropriate PSAP (according to ECRF).

5. SIP server at PSAP sends reply (200) to ESRP SIP Proxy. Message contains SDP (WebRTC Answer).

6. SIP client gets SDP (Answer) and sends ACK back. Both parties now have all data needed to establish WebRTC connection.

7. WebRTC client at PWA establishes RTP connection with WebRTC client at PSAP (Sfera). Audio/video and data (instant messaging) is transmitted via TURN server.

Having the PWA + SIP client in JavaScript makes it possible for the caller to use the application even if the SOS website is temporarily unavailable.

Another benefit of using a JavaScript SIP client is in using WebSocket for communication between SIP UA and ESRP (RFC 7118). This simplifies the Border Control Function (BCF) which can be the standard reverse proxy which is normally used for public websites. It also enables the use of standard web secure communication (wss – WebSocket Secure) over standard port 443. There is no need to open standard (and not encrypted) 5060 SIP port.

The TURN server must be “visible” to both the caller and the operator in the PSAP. It is installed outside of ESInet and the network at both sides should be configured to allow the TCP/UDB connection to port 3478 and UDP to port range 49152-65535. This basically means that the proxy/firewall on the PSAP’s side should be set up to allow connection through these ports. On the caller’s side, there should be no need for any configuration if the call is made by a smartphone or by a laptop behind a home ADSL modem. If the caller is behind a symmetric firewall (e.g. inside company), then ports should also be enabled on the caller’s side.

Text communication (i.e. chat) is carried out using RTCDataChannel, which is part of the WebRTC standard. This avoids problems with routing text messages back and forth if MESSAGE SIP is used for text message exchange, which is also a possibility. Initially, text message communication was implemented with standard SIP MESSAGES, but we found that using WebRTC (RTCDataChannel) is a more practical solution. Text "call" is established only once and the caller is guaranteed to stay with the same operator in the PSAP during the entire text call session. Also, ESRP and the entire SIP infrastructure is far less utilised for the single text call session.

ESRP service is implemented using the *SIPSorcery library*[2].

For JavaScript, the *SIP jsSIP library*[3] is used.

For TURN, the *server coturn*[4] is used.

All other parts of the system are implemented using C#, Javascript, .NET and MS SQL tehnologies.

Figure 5 shows communication and messaging protocols used between system components.

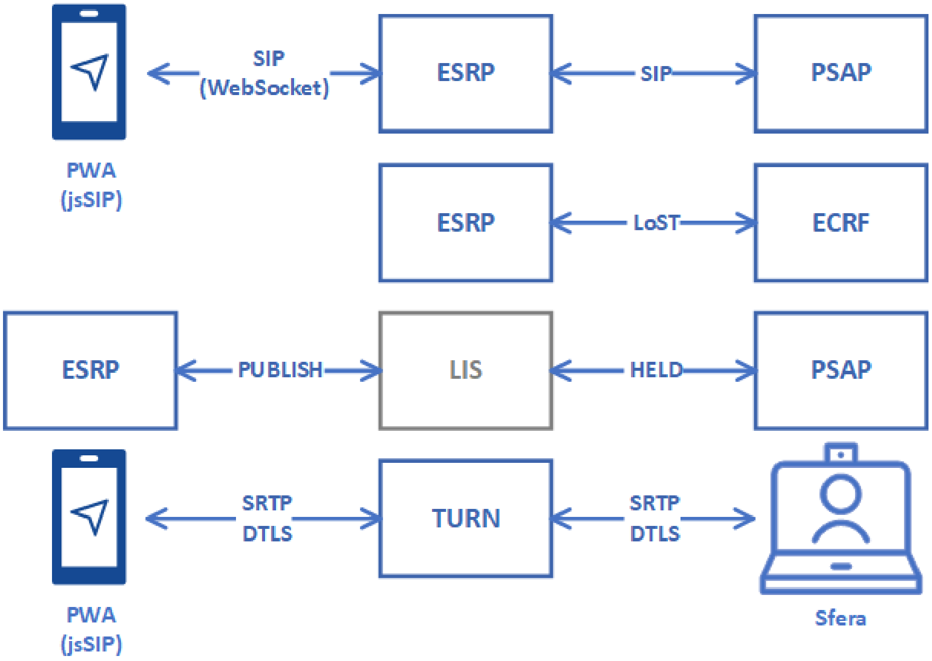


Figure 5: Protocols used in solution

[2] *https://github.com/sipsorcery/sipsorcery*
[3] *https://github.com/versatica/JsSIP*
[4] *https://github.com/coturn/coturn*

Figure 6 shows a more detailed call sequence between the caller and the PSAP.

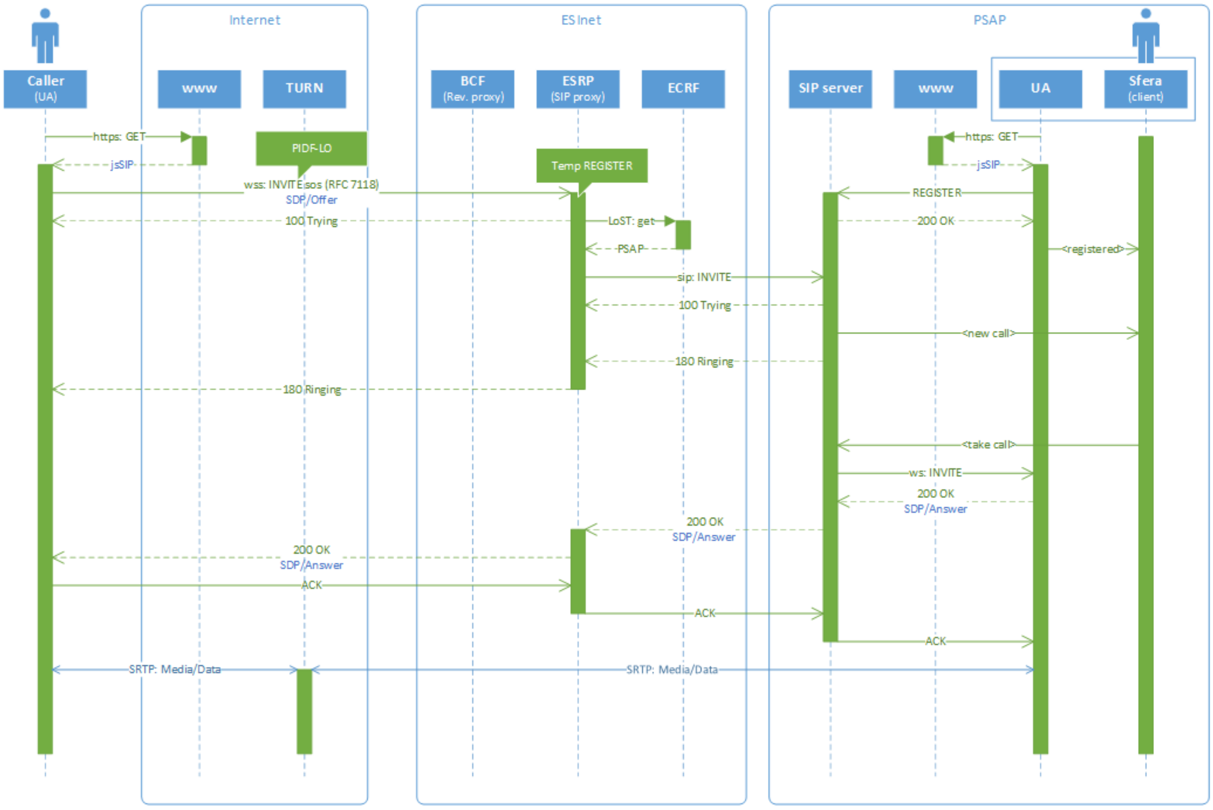


Figure 6: SIP SOS call sequence

- The caller initially loads the web application from the SOS website, or just starts it from the home screen if it is installed locally. If it is the first time the application is opened, the caller should enter their own phone number and name so that the PSAP can call them back should the need arise later.

- When the caller selects video, audio or text SOS call, the JavaScript SIP client connects to ESRP using secure WebSockets (wss://) and sends the INVITE SIP message.The URL of ESRP is given by the SOS website and saved locally. In the INVITE message, PIDF-LO message and SDP Offer are sent as multipart/mixed content. UA maintain WebSocket connection with ESRP during the entire call session.

- ESRP validates the caller for mandatory data (phone number, name, PDIF-LO, SDP) and temporary registers it in the memory. The registration is active as the WebSocket connection is maintained. This enables the ESRP to route back SIP messages to the caller UA.

- ESRP extracts the PDIF-LO message and uses it to call the ECRF service using LoST protocol.
- ECRF returns the most appropriate PSAP. In this project, the appropriate PSAP is determined by caller geolocation. In a production-ready solution, more rules could be implemented, such as type of call, load status and/or working hours of PSAPs etc.
- ESRP forwards INVITE message to next hop, i.e. next ESRP or destination PSAP. In this project, the message is forwarded to the appropriate PSAP.
- The SIP server at the PSAP site, which is integrated with Sfera CAD system, receives the INVITE message and notifies the Sfera system that there is new call.
- Sfera puts the call in the PSAP waiting queue and alerts the logged in operators through the Sfera application.
- In the meantime, the 112 operator has registered with the PSAP's SIP server, also using jsSIP SIP UA.
- SIP server returns the 180 Ringing message back to the ESRP.
- ESRP forwards the 180 Ringing message back to the caller.
- The first available operator in the PSAP takes the new call.
- The SIP server forwards the INVITE message to the operater's UA.
- The operator's UA takes the SDP Offer from the message, constructs the SDP Answer and sends back the 200 OK message. At the same time, the case is opened in the Sfera application with caller location, name and phone number displayed.
- The 200 OK message traverses back to the caller.
- The caller's UA takes the SDP Answer and uses it to establish a connection with the operator at the 112 PSAP.
- Both the caller's and operator's WebRTC clients now have all the data needed to establish the SRTP connection via TURN server.
- Depending on the call type, media streams (audio, video) or just text data (via RTCDataChannel) are transferred in both directions via TURN server.

5 | TESTS

One of objectives of this project was to integrate the NG112 solution into our CAD system – Sfera – which is implemented in 15 out of 20 112 PSAPs in Croatia. Testing was done on the Sfera testing environment in two regions (counties) in Croatia:

1 – Zagreb county (Zagrebačka županija) and

21 – City of Zagreb (Grad Zagreb)

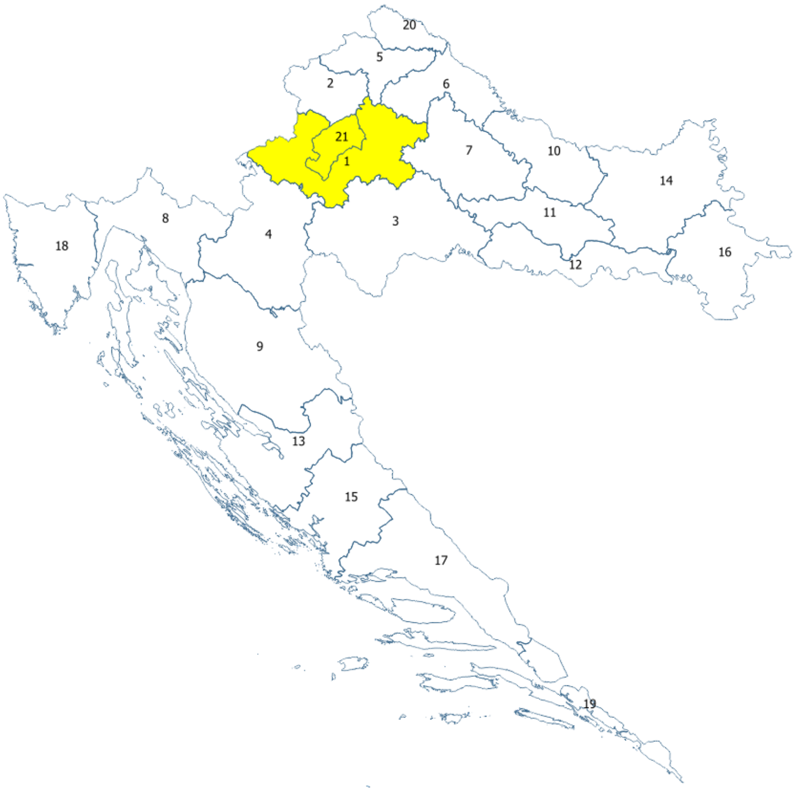
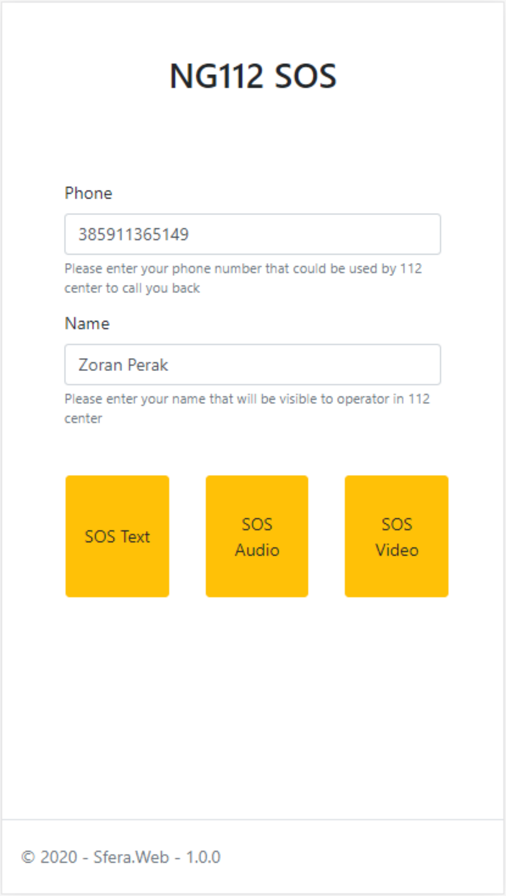


Figure 7: Regions in test

We used geo-shapes from production which also contain shapes of counties that ECRF uses to determine the most appropriate PSAP.

The domain used for testing is ng112.net (website at *https://sos.ng112.net*).

Figure 8 shows the home screen of the client application. The caller enters their own phone number and name in the input fields and initiates the SOS call by pressing the appropriate button.



NG112 SOS

Phone

385911365149

Please enter your phone number that could be used by 112 center to call you back

Name

Zoran Perak

Please enter your name that will be visible to operator in 112 center

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Figure 8: PWA initial screen

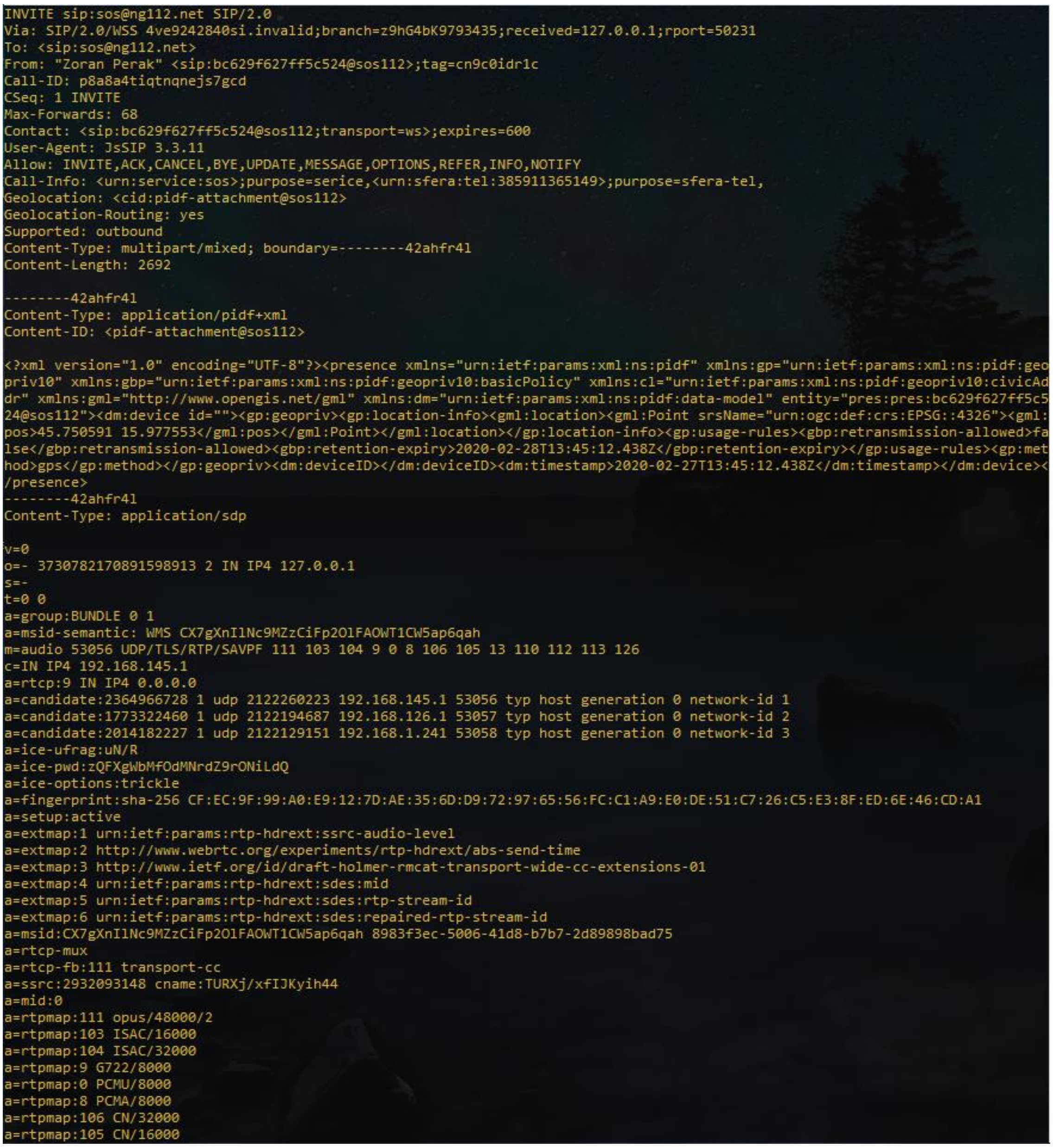
Regardless of the type of SOS call, the SIP client connects to the ESRP server via the WebSocket (wss) and sends the INVITE message to the ESRP.

Figure 9 shows a message example. The caller's name is sent as "Display Name" in the "From" header. The caller's phone number is sent via "Call-Info" header in the application's custom "urn:sfera:tel" tag inside the "Caller-Info" header. Unfortunately, there is no standard tag that can be used specifically for this purpose. Sent in the same header is the "standard" urn for emergency calls: "urn:service:sos". Unfortunately, again, the jsSIP client does not support sending the INVITE message to such urn.

Geolocation is sent with the INVITE message as part of the multipart content. Another part is the SDP Offer message that contains information of all relevant capabilities, depending on the call type (supported codecs, media capabilities…).

SDP also contains ICE candidates, especially TURN server data since it is not probable that the caller would be able to establish a media stream with the operator in the PSAP without usage of the TURN server.

To make things simpler, we have decided to collect all SDP data upfront, meaning that the initial INVITE message from the caller would contain all the information from the caller's side that will be used on the PSAP's side to establish the media or data connection. The same holds for the PSAP's side. The Complete SDP Answer is sent with the 200 OK message from the PSAP to the caller.



```
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To: <sip:sos@ng112.net>
From: "Zoran Perak" <sip:bc629f627ff5c524@sos112>;tag=cn9c0idr1c
Call-ID: p8a8a4tiqtnqnejs7gcd
CSeq: 1 INVITE
Max-Forwards: 68
Contact: <sip:bc629f627ff5c524@sos112;transport=ws>;expires=600
User-Agent: JsSIP 3.3.11
Allow: INVITE,ACK,CANCEL,BYE,UPDATE,MESSAGE,OPTIONS,REFER,INFO,NOTIFY
Call-Info: <urn:service:sos>;purpose=serice,<urn:sfera:tel:385911365149>;purpose=sfera-tel,
Geolocation: <cid:pidf-attachment@sos112>
Geolocation-Routing: yes
Supported: outbound
Content-Type: multipart/mixed; boundary=--------42ahfr41
Content-Length: 2692

--------42ahfr41
Content-Type: application/pidf+xml
Content-ID: <pidf-attachment@sos112>

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/presence>
--------42ahfr41
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s=-
t=0 0
a=group:BUNDLE 0 1
a=msid-semantic: WMS CX7gXnIlNc9MZzCiFp2OlFAOWT1CW5ap6qah
m=audio 53056 UDP/TLS/RTP/SAVPF 111 103 104 9 0 8 106 105 13 110 112 113 126
c=IN IP4 192.168.145.1
a=rtcp:9 IN IP4 0.0.0.0
a=candidate:2364966728 1 udp 2122260223 192.168.145.1 53056 typ host generation 0 network-id 1
a=candidate:1773322460 1 udp 2122194687 192.168.126.1 53057 typ host generation 0 network-id 2
a=candidate:2014182227 1 udp 2122129151 192.168.1.241 53058 typ host generation 0 network-id 3
a=ice-ufrag:uN/R
a=ice-pwd:zQFXgWbMfOdMNrdZ9rONiLdQ
a=ice-options:trickle
a=fingerprint:sha-256 CF:EC:9F:99:A0:E9:12:7D:AE:35:6D:D9:72:97:65:56:FC:C1:A9:E0:DE:51:C7:26:C5:E3:8F:ED:6E:46:CD:A1
a=setup:active
a=extmap:1 urn:ietf:params:rtp-hdrext:ssrc-audio-level
a=extmap:2 http://www.webrtc.org/experiments/rtp-hdrext/abs-send-time
a=extmap:3 http://www.ietf.org/id/draft-holmer-rmcat-transport-wide-cc-extensions-01
a=extmap:4 urn:ietf:params:rtp-hdrext:sdes:mid
a=extmap:5 urn:ietf:params:rtp-hdrext:sdes:rtp-stream-id
a=extmap:6 urn:ietf:params:rtp-hdrext:sdes:repaired-rtp-stream-id
a=msid:CX7gXnIlNc9MZzCiFp2OlFAOWT1CW5ap6qah 8983f3ec-5006-41d8-b7b7-2d89898bad75
a=rtcp-mux
a=rtcp-fb:111 transport-cc
a=ssrc:2932093148 cname:TURXj/xfIJKyih44
a=mid:0
a=rtpmap:111 opus/48000/2
a=rtpmap:103 ISAC/16000
a=rtpmap:104 ISAC/32000
a=rtpmap:9 G722/8000
a=rtpmap:0 PCMU/8000
a=rtpmap:8 PCMA/8000
a=rtpmap:106 CN/32000
a=rtpmap:105 CN/16000
```
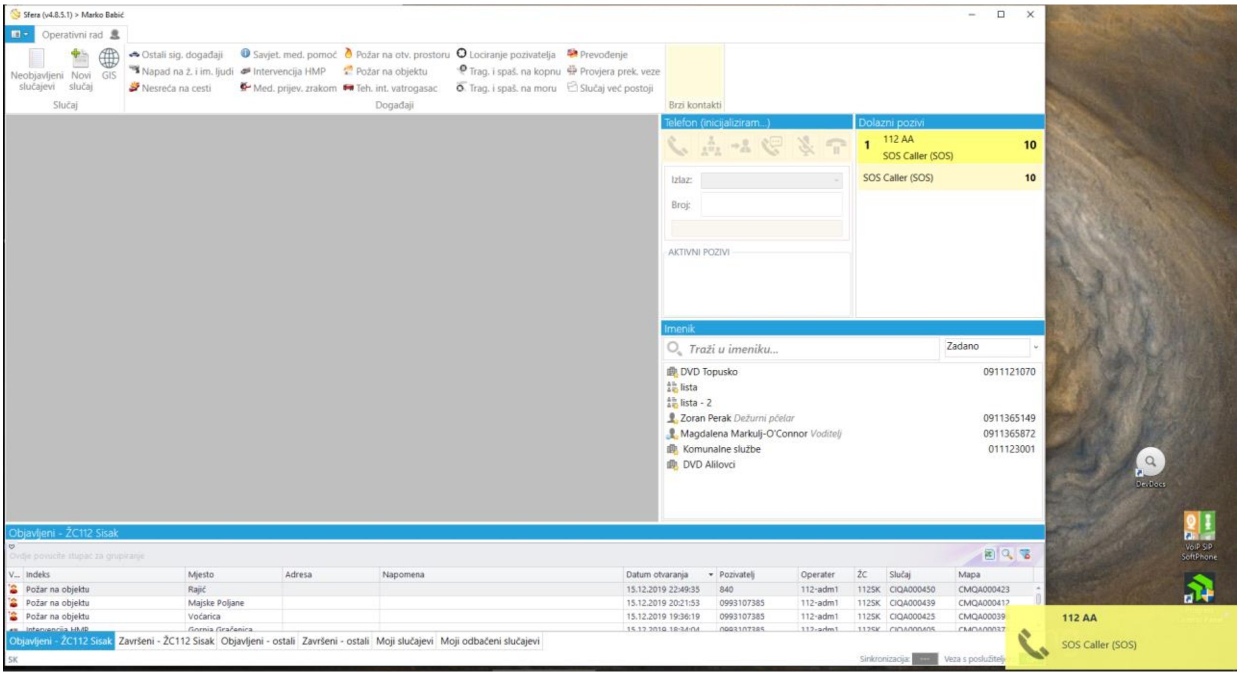

Figure 9: INVITE message from caller

When the INVITE message arrives to the appropriate PSAP, the call is put into a waiting queue and the operator is alerted accordingly as shown on *Figure 10*.



Dolazni pozivi

1	112 AA SOS Caller (SOS)	10
	SOS Caller (SOS)	10

Imenik

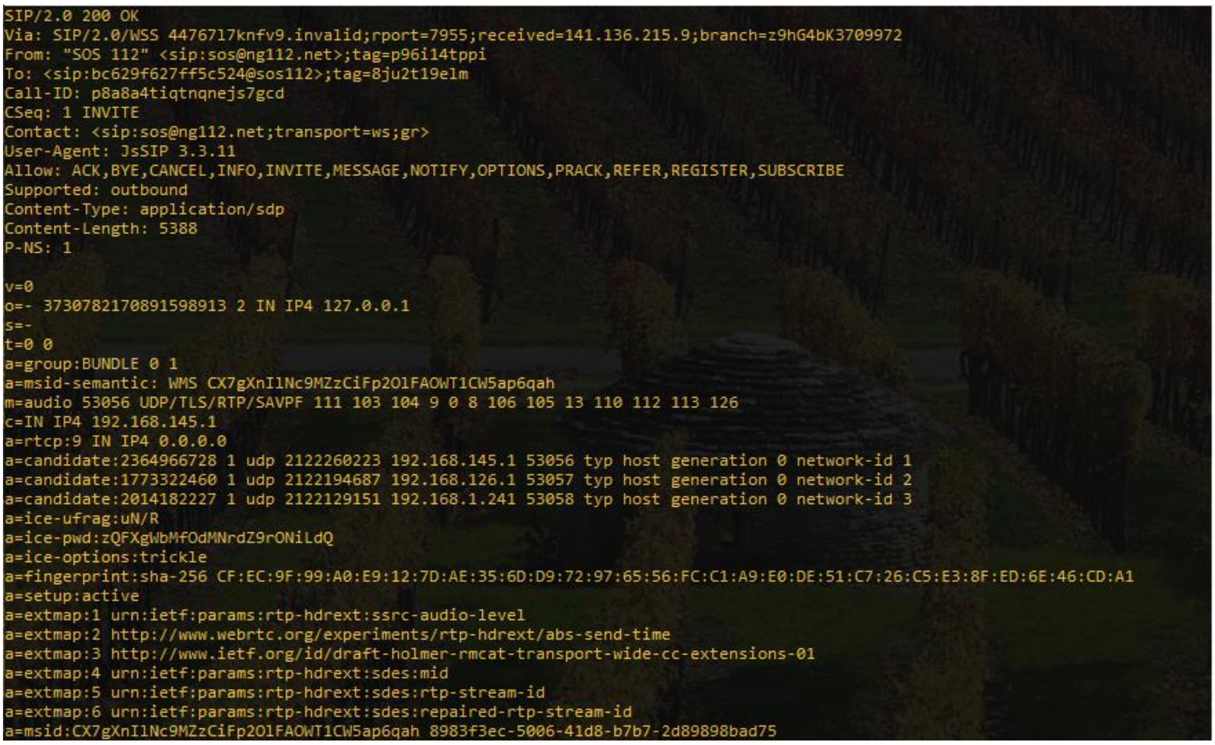
DVD Topusko	0911121070
lista	
lista - 2	
Zoran Perak *Dežurni pčelar*	0911365149
Magdalena Markulj-O'Connor *Voditelj*	0911365872
Komunalne službe	011123001
DVD Alilovci	

V...	Indeks	Mjesto	Adresa	Napomena	Datum otvaranja	Pozivatelj	Operater	ŽC	Slučaj	Mapa
	Požar na objektu	Rajić			15.12.2019 22:49:35	840	112-adm1	112SK	CIQA000450	CMQA000423
	Požar na objektu	Majske Poljane			15.12.2019 20:21:53	0993107385	112-adm1	112SK	CIQA000439	CMQA000412
	Požar na objektu	Voćarica			15.12.2019 19:36:19	0993107385	112-adm1	112SK	CIQA000425	CMQA00039
	Intervencija HMP	Gornja Gračenica			15.12.2019 18:34:04	0993107385	112-adm1	112SK	CIQA000405	CMQA00037

112 AA
SOS Caller (SOS)

Figure 10: Sfera (PSAP system) receiving web call

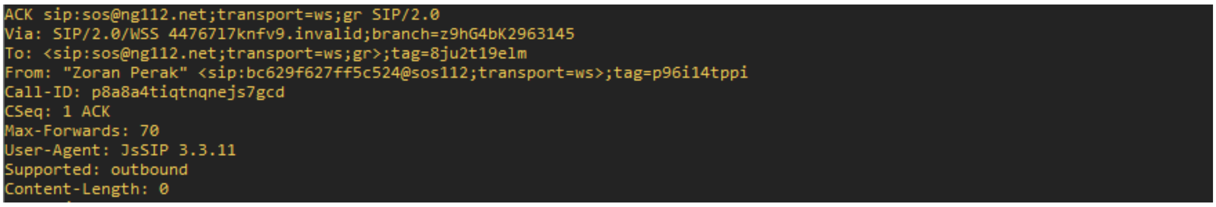
When the operator in the PSAP takes the incoming call, in that moment the SIP client (jsSIP) on her/his computer receives the original INVITE message from the caller. The SDP is then extracted from the message and supplied to the local WebRTC engine which produces the Answer SDP in return. The SDP is also supplied with TURN data from the PSAP’s side and sent back to the caller in the200 OK message (see *Figure 11)*.



```
SIP/2.0 200 OK
Via: SIP/2.0/WSS 4476717knfv9.invalid;rport=7955;received=141.136.215.9;branch=z9hG4bK3709972
From: "SOS 112" <sip:sos@ng112.net>;tag=p96i14tppi
To: <sip:bc629f627ff5c524@sos112>;tag=8ju2t19elm
Call-ID: p8a8a4tiqtnqnejs7gcd
CSeq: 1 INVITE
Contact: <sip:sos@ng112.net;transport=ws;gr>
User-Agent: JsSIP 3.3.11
Allow: ACK,BYE,CANCEL,INFO,INVITE,MESSAGE,NOTIFY,OPTIONS,PRACK,REFER,REGISTER,SUBSCRIBE
Supported: outbound
Content-Type: application/sdp
Content-Length: 5388
P-NS: 1

v=0
o=- 3730782170891598913 2 IN IP4 127.0.0.1
s=-
t=0 0
a=group:BUNDLE 0 1
a=msid-semantic: WMS CX7gXnIlNc9MZzCiFp2OlFAOWT1CW5ap6qah
m=audio 53056 UDP/TLS/RTP/SAVPF 111 103 104 9 0 8 106 105 13 110 112 113 126
c=IN IP4 192.168.145.1
a=rtcp:9 IN IP4 0.0.0.0
a=candidate:2364966728 1 udp 2122260223 192.168.145.1 53056 typ host generation 0 network-id 1
a=candidate:1773322460 1 udp 2122194687 192.168.126.1 53057 typ host generation 0 network-id 2
a=candidate:2014182227 1 udp 2122129151 192.168.1.241 53058 typ host generation 0 network-id 3
a=ice-ufrag:uN/R
a=ice-pwd:zQFXgWbMfOdMNrdZ9rONiLdQ
a=ice-options:trickle
a=fingerprint:sha-256 CF:EC:9F:99:A0:E9:12:7D:AE:35:6D:D9:72:97:65:56:FC:C1:A9:E0:DE:51:C7:26:C5:E3:8F:ED:6E:46:CD:A1
a=setup:active
a=extmap:1 urn:ietf:params:rtp-hdrext:ssrc-audio-level
a=extmap:2 http://www.webrtc.org/experiments/rtp-hdrext/abs-send-time
a=extmap:3 http://www.ietf.org/id/draft-holmer-rmcat-transport-wide-cc-extensions-01
a=extmap:4 urn:ietf:params:rtp-hdrext:sdes:mid
a=extmap:5 urn:ietf:params:rtp-hdrext:sdes:rtp-stream-id
a=extmap:6 urn:ietf:params:rtp-hdrext:sdes:repaired-rtp-stream-id
a=msid:CX7gXnIlNc9MZzCiFp2OlFAOWT1CW5ap6qah 8983f3ec-5006-41d8-b7b7-2d89898bad75
```

Figure 11: 200 OK message from PSAP to caller

After reception of the 200 OK message, the caller extracts the SDP and feeds it to the local WebRTC engine. Both WebRTC engines now have all the data needed to establish the media connection and start streaming. To acknowledge that all is OK, the caller sends an ACK message to the PSAP (see *Figure 12)*.



```
ACK sip:sos@ng112.net;transport=ws;gr SIP/2.0
Via: SIP/2.0/WSS 4476717knfv9.invalid;branch=z9hG4bK2963145
To: <sip:sos@ng112.net;transport=ws;gr>;tag=8ju2t19elm
From: "Zoran Perak" <sip:bc629f627ff5c524@sos112;transport=ws>;tag=p96i14tppi
Call-ID: p8a8a4tiqtnqnejs7gcd
CSeq: 1 ACK
Max-Forwards: 70
User-Agent: JsSIP 3.3.11
Supported: outbound
Content-Length: 0
```

Figure 12: ACK message from caller to PSAP

Finally, WebRTC media streams are established and presented to both the caller and the operator in the PSAP. On the caller's side, the rear camera is turned on by default and a bigger video is presented on the screen from that camera. The video of the operator is presented in a smaller frame. This is different from the usual way that a video call is presented. The reason for that is that the most likely point of interest is the caller's surrounding rather than the caller's face and the caller should direct the rear camera to the place of incident. Therefore, the video from his/her camera is presented over the entire screen.

The video from the PSAP (the operator's face) is most likely less important. During the video call, the sound is turned to speakers so that the caller can normally communicate with the operator via speakerphone (see *Figure 13)*.

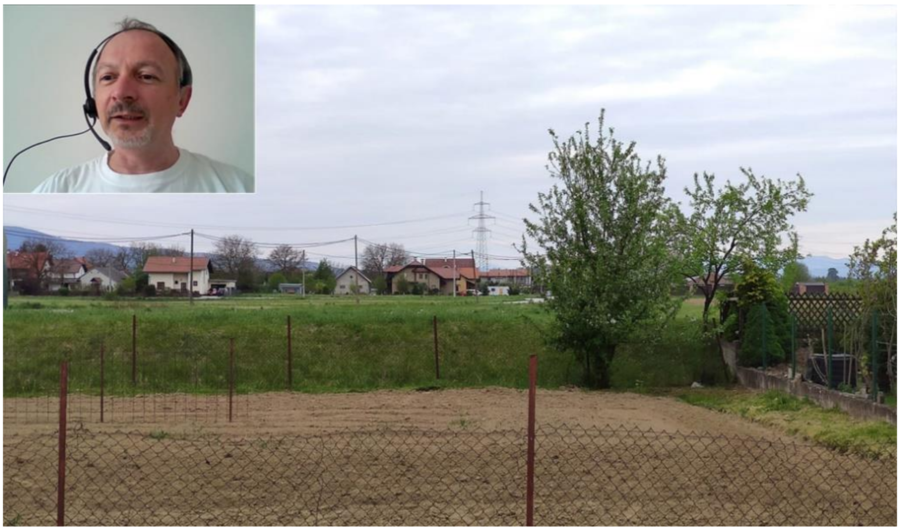
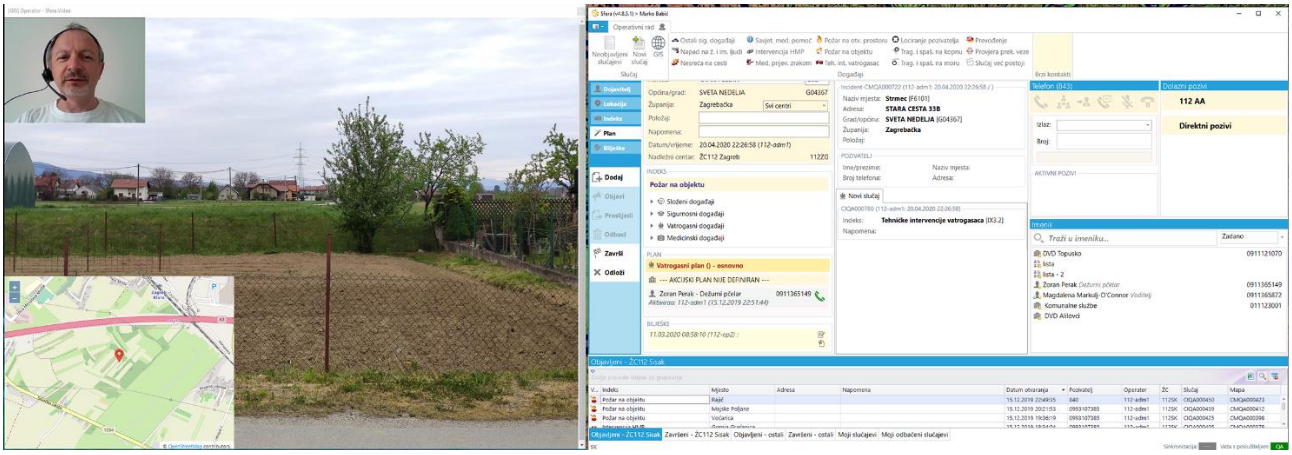


Figure 13: PWA during video call

On the PSAP's side, the same composition is used, i.e. a bigger video from the caller's rear camera and a smaller video of the operator, plus a map with the live location of the caller (see *Figure 14)*.

The web call is integrated with the Sfera system so the operator can use other "regular" functionalities to handle the incident.



Općina/grad: SVETA NEDELJA G04367
Županija: Zagrebačka Svi centri
Položaj:
Napomena:
Datum/vrijeme: 20.04.2020 22:26:58 (112-adm1)
Nadležni centar: ŽC112 Zagreb 112ZG

INDEKS
Požar na objektu
- Složeni događaji
- Sigurnosni događaji
- Vatrogasni događaji
- Medicinski događaji

PLAN
Vatrogasni plan () - osnovno
--- AKCIJSKI PLAN NIJE DEFINIRAN ---
Zoran Perak - Dežurni pčelar 0911365149
Aktivirao: 112-adm1 (15.12.2019 22:51:44)

BILJEŠKE
11.03.2020 08:58:10 (112-op2) :

Incident-CMQA000722 (112-adm1: 20.04.2020 22:26:58 /)
Naziv mjesta: Strmec [F6101]
Adresa: STARA CESTA 33B
Grad/općina: SVETA NEDELJA [G04367]
Županija: Zagrebačka
Položaj:

POZIVATELJ
Ime/prezime: Naziv mjesta:
Broj telefona: Adresa:

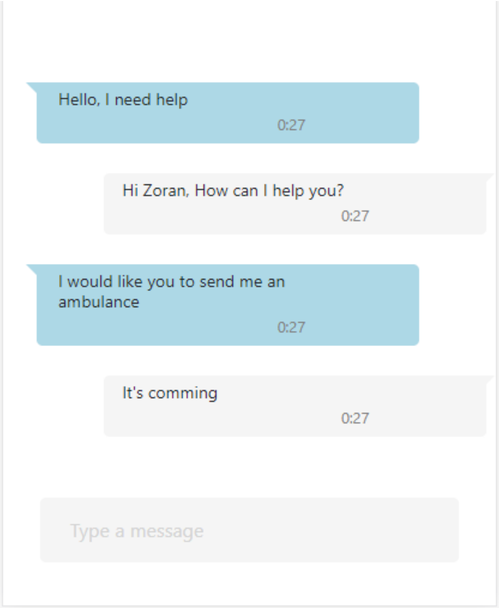
Novi slučaj
CIQA000780 (112-adm1: 20.04.2020 22:26:58)
Indeks: Tehničke intervencije vatrogasaca [IX3.2]
Napomena:

DVD Topusko	0911121070
lista	
lista - 2	
Zoran Perak Dežurni pčelar	0911365149
Magdalena Markulj-O'Connor Voditelj	0911365872
Komunalne službe	011123001
DVD Alilovci	

V...	Indeks	Mjesto	Adresa	Napomena	Datum otvaranja	Pozivatelj	Operater	ŽC	Slučaj	Mapa
	Požar na objektu	Rajić			15.12.2019 22:49:35	840	112-adm1	112SK	CIQA000450	CMQA000423
	Požar na objektu	Majske Poljane			15.12.2019 20:21:53	0993107385	112-adm1	112SK	CIQA000439	CMQA000412
	Požar na objektu	Vočarica			15.12.2019 19:36:19	0993107385	112-adm1	112SK	CIQA000425	CMQA000398

Figure 14: Sfera (PSAP) during video call. Screenshot from two monitors.

Figure 15 shows the smartphone client during a text call session and *Figure 16* shows the PSAP workstation during the same session. The text call session was taken on the same location as the video call. The chat widget and map are implemented in rudimentary form. For production use, a more polished version should be built.



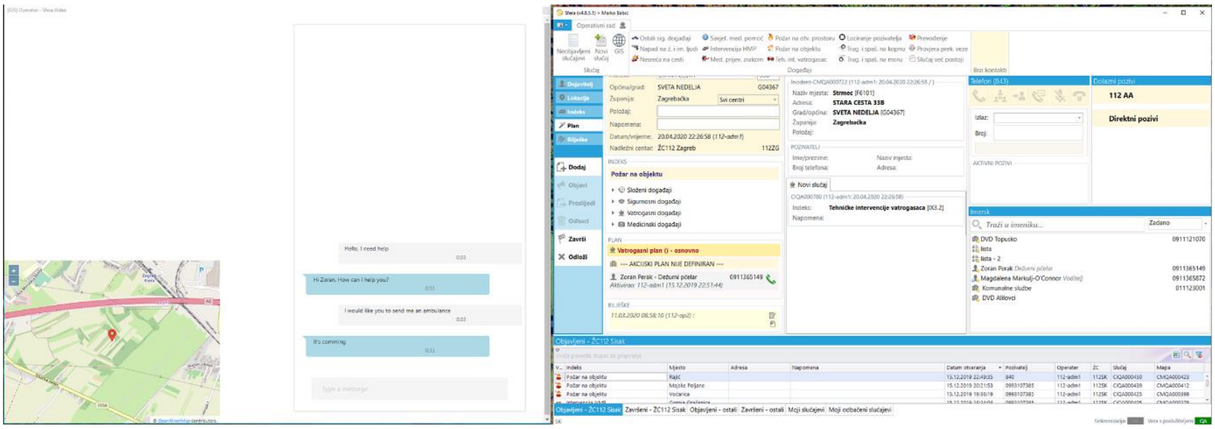
Hello, I need help
0:27

Hi Zoran, How can I help you?
0:27

I would like you to send me an ambulance
0:27

It's comming
0:27

Figure 15: PWA during text call (chat)



Hello, I need help

Hi Zoran, How can I help you?

I would like you to send me an ambulance

It's comming

Figure 16: Sfera (PSAP) during video call

6 | LESSONS LEARNT & RECOMMENDATIONS

The key benefit of the proposed solution is that the person in distress does not need to have pre-installed any special mobile application to initiate the emergency video call. It uses a standard web browser already available on practically any device. Of course, the URL of the SOS site should be widely known and publicised, like the 112 number.

The video stream from the accident site is of great value for emergency services to accurately assess the situation and engage the most appropriate resources.

Using PWA + JavaScript SIP client gives us benefits:

- Works even if SOS site is temporarily down
- Using od SIP over WebSockets make deployment and security consideration much easier

Some implementation challenges, and in part thanks to comments from colleagues from the CELESTE consortium team, made us rethink the text call solution. Initially we implemented it by sending the MESSAGE SIP message for each text piece transferred between the caller and the PSAP. That poses a problem of maintaining the connection with the same PSAP and the same operator during the text chat session, even if the caller moves and enters an area covered by another PSAP. This was solved by using Record-Route headers during the initial message path from the caller to the PSAP and later using the recorded route with all other messages, effectively forcing the infrastructure to route each message to the same path. However, that seemed a bit unnecessarily complicated and also presented a certain burden on the infrastructure, which must route every message.

In the end, we decided to re-implement it by using the WebRTC data channel instead. It proved to be a relatively easy code refactoring.

6.1 | MAIN CONSIDERATIONS

Anonymous vs registered access to 112

It is up to the PSAP authorities to decide whether it will allow anonymous calls to the 112 emergency services. Allowing anonymous calls could add "noise" to PSAP operations because it is more tempting for some people to make invalid calls to emergency services for fun or for malice if they think that these calls are untraceable.

On the other hand, making it harder and more complicated for people to call emergency services when they are in distress and have no time to waste negates the purpose of the service.

An acceptable compromise between the two could be in asking the caller to voluntarily enter their own phone number or to make "registration" very simple:

- Caller enters their own phone number and taps the button "Register"
- The web server sends an SMS message to that phone number with link for account activation
- The caller just opens the link in the received SMS message and after that her/his SOS application is registered and she/he can proceed to call 112

Internet access – speed

Although mobile internet access is constantly gaining in speed and coverage, it is still likely to enter an area with very slow or no internet connection at all.

This can be mitigated with offering the caller only audio and text call if internet speed is detected to be slow or just text call if it is very slow.

NG112 is not meant to replace the existing phone call system but rather to supplement it. If there is no internet connection, it is probably still possible to make regular phone call.

Internet access – costs

Mobile internet access is ubiquitous and everyone has it. Is this really so? When using a smartphone it's not always easy to be able to control internet consumption, which often leads to situations where they use all their available gigabytes well before new gigabytes arrive the next month. That leaves us with the not unlikely possibility that a person who needs to call SOS in that moment might not have internet available. Also, some foreign citizens might find internet roaming prices prohibitive to use.

This situation can probably only be resolved by legislators by requiring mobile telecom operators to make internet traffic to certain domains free of charge, similarly to free phone calls to 112.

7 | CONCLUSIONS & NEXT STEPS

This project proved that NG112 technology is available, ready and viable to be implemented today and not in some uncertain future.

The next logical step could be to pilot this project in production. In the first phase, ESInet would be established together with a single PSAP that would cover the whole country, but only for NG112 calls. In this pilot project, invaluable experience would be accumulated that would enable refinement and improvement of the system.

In the next phase, one by one, PSAPs would be connected to the ESInet, upgraded to be NG112 compatible and would start to also handle NG112 calls. During the transition period, the "main" NG112 PSAP would cover less and less area until all PSAPs would become NG112 able.

8 | TERMS & ABBREVIATIONS

BCF	Border Control Function (Firewall, on PSAP side not used in demo)
ECRF	Emergency Call Routing Function
ESInet	Emergency Services IP Network
ESRP	Emergency Service Routing Proxy
LIS	Location Information Service
LoST	Location to Service Translation
PDIF-LO	Presence Information Data Format-Location Object
PSAP	Public Safety Answering Point
PSTN	Public Switched Telephone Network
PWA	Progressive Web Application
RFC	Request For Comment
SDP	Session Description Protocol
Sfera	PSAP CAD (Computer Aided Dispatch) system from KING ICT
SIP	Session Initiation Protocol
SRTP	Secure Real-time Transport Protocol
TURN	Traversal Using Relay NAT
UA	User Agent
WebRTC	Web Real-Time Communication