



# GÉANT WebRTC Roadmap

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- WebRTC and Unified Communications
- Contextual communication
- NAT/Firewall traversal (STUN/TURN)
- WebTUT application demo

# WebRTC is coming, like it or not...

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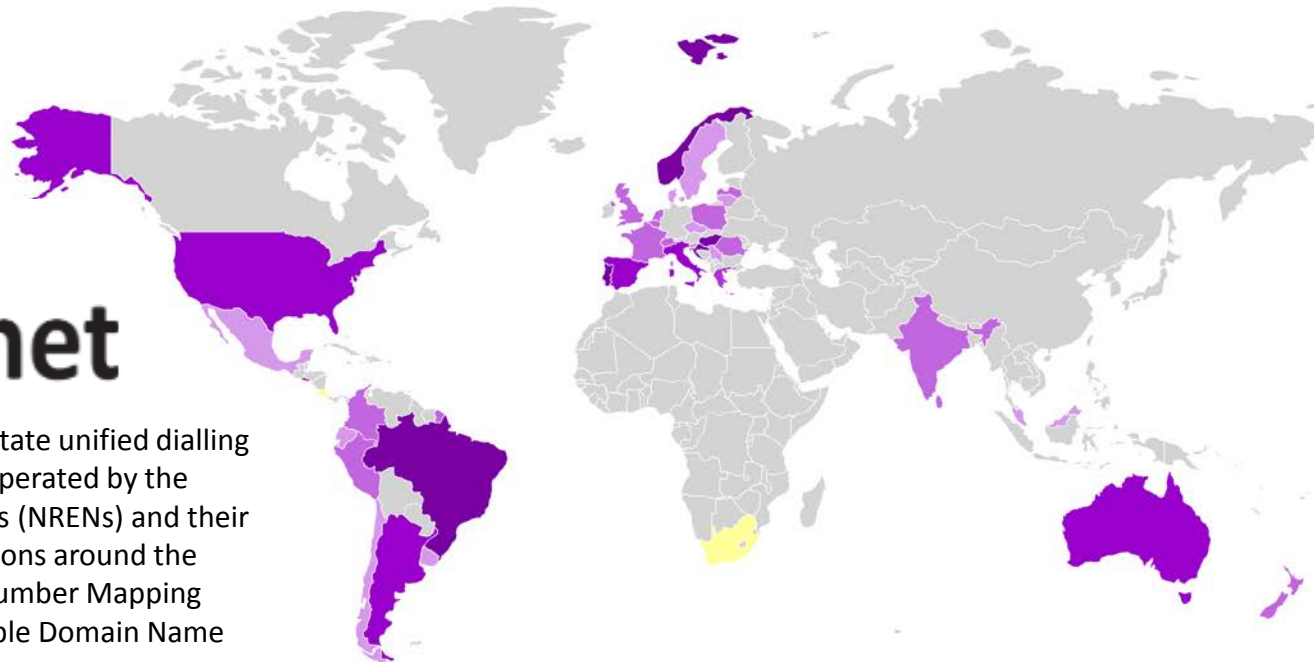
**Technology, not a turn key solution!**

# Supporting Unified Communications

- Unified Communication (UC) is sufficiently mature to replace telephone systems at R&E institutions. Solutions from Microsoft and Cisco, in particular, are offered at very favourable conditions to the R&E community, complemented with cloud delivery models that lower deployment costs significantly.
- Different national communities are at different points in this transition.



Global service, initiated by GÉANT, to facilitate unified dialling of voice and video conferencing services operated by the National Research and Education Networks (NRENs) and their connected academic and research institutions around the world. It is based on the standard E.164 Number Mapping (ENUM) protocol and the secure and reliable Domain Name Server (DNS) infrastructure of NRENs.



top country codes

#	country name	E.164	ENUMs
1.)	 Hungary	<a href="#">+36</a>	79715
2.)	 Norway	<a href="#">+47</a>	67859
3.)	 Portugal	<a href="#">+351</a>	49006
4.)	 Brazil	<a href="#">+55</a>	10048
5.)	 Spain	<a href="#">+34</a>	6904
6.)	 North American Numbering Plan	<a href="#">+1</a>	5035
7.)	 Argentina	<a href="#">+54</a>	3462
8.)	 Australia	<a href="#">+61</a>	1946
9.)	 Italy	<a href="#">+39</a>	932
10.)	 Greece	<a href="#">+30</a>	911
11.)	 Croatia	<a href="#">+385</a>	612
12.)	 El Salvador	<a href="#">+503</a>	102
13.)	 United Kingdom	<a href="#">+44</a>	49
14.)	 India	<a href="#">+91</a>	22
15.)	 Netherlands	<a href="#">+31</a>	22
16.)	 Latvia	<a href="#">+371</a>	21
17.)	 Czech Republic	<a href="#">+420</a>	20
18.)	 Hong Kong	<a href="#">+852</a>	20
19.)	 Belgium	<a href="#">+32</a>	11
20.)	 New Zealand	<a href="#">+64</a>	10
21.)	 Sri Lanka	<a href="#">+94</a>	10
22.)	 France	<a href="#">+33</a>	8
23.)	 Poland	<a href="#">+48</a>	1
24.)	 Romania	<a href="#">+40</a>	1
25.)	 Chile	<a href="#">+56</a>	1

## WebRTC and Unified Communications

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- If we assume that UC solutions will be present at most institutions in European R&E some years from now, they will represent the legacy technology WebRTC solutions are most likely to meet.
  - At the moment, both UC and WebRTC solutions are being developed and deployed. This makes it very timely to investigate how the two play together and whether there are smart moves that R&E institutions should make now to support a smooth and effective RTC infrastructure in the near future.
1. An obvious, but nonetheless important, improvement WebRTC brings to UC, is improved guest access and easier deployment to licensed users. It enables UC vendors to create a web-based client that offers the same quality audio and video as native clients.
  2. Less obvious is WebRTC's capability to de-unify Unified Communications. When people work together, communication is contextual.

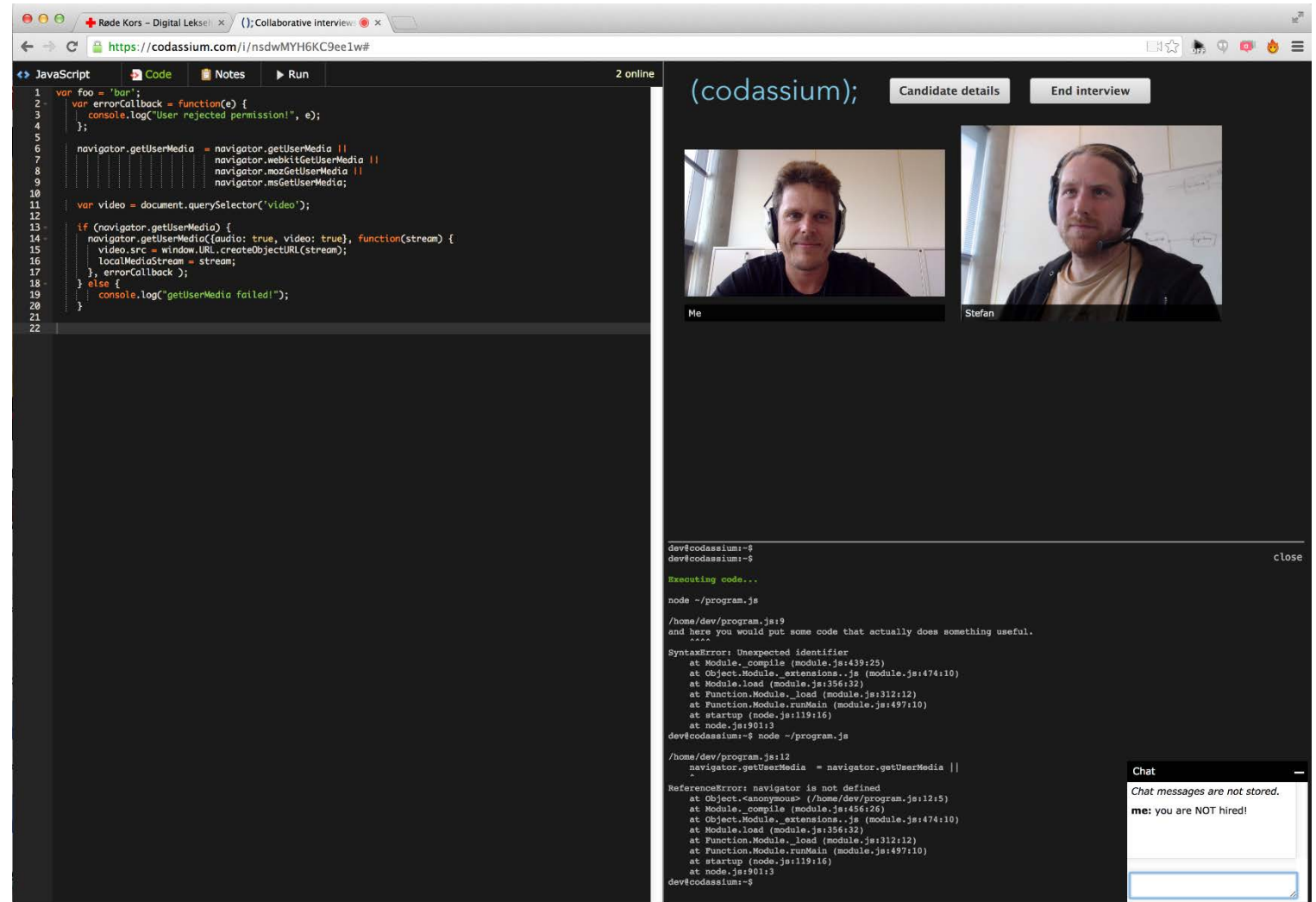
# Contextual communication

- **Current situation:**

- you work on X in product Y
- you talk together in product Z

- **In-context communication:**

- you work on X in product Y
- you talk together in product Y



The screenshot displays a web-based collaborative interview environment. On the left, a code editor shows JavaScript code for a video player. The main area features two video feeds: 'Me' and 'Stefan'. Below the videos is a chat window with a message: 'me: you are NOT hired!'. The interface includes a terminal at the bottom showing a syntax error and a reference error.

```
1 var foo = "bar";
2 var errorCallback = function(e) {
3   console.log("User rejected permission!", e);
4 };
5
6 navigator.getUserMedia = navigator.getUserMedia ||
7 navigator.webkitGetUserMedia ||
8 navigator.mozGetUserMedia ||
9 navigator.msGetUserMedia;
10
11 var video = document.querySelector("video");
12
13 (function(navigator.getUserMedia) {
14   navigator.getUserMedia({audio: true, video: true}, function(stream) {
15     video.src = window.URL.createObjectURL(stream);
16     localMediaStream = stream;
17   }, errorCallback);
18 } else {
19   console.log("getUserMedia failed!");
20 }
21
22
```

dev@codassium:~\$ node ~/program.js  
SyntaxError: Unexpected identifier  
at Module.compile (module.js:439:25)  
at Object.Module.\_extensions..js (module.js:474:10)  
at Module.load (module.js:356:32)  
at Function.Module.\_load (module.js:312:12)  
at Function.Module.runMain (module.js:497:10)  
at startup (node.js:119:16)  
at node.js:901:3  
dev@codassium:~\$ node ~/program.js  
ReferenceError: navigator is not defined  
at Object.<anonymous> (/home/dev/program.js:12:5)  
at Module.compile (module.js:456:26)  
at Object.Module.\_extensions..js (module.js:474:10)  
at Module.load (module.js:356:32)  
at Function.Module.\_load (module.js:312:12)  
at Function.Module.runMain (module.js:497:10)  
at startup (node.js:119:16)  
at node.js:901:3  
dev@codassium:~\$

# STUN/TURN Technology Scouting

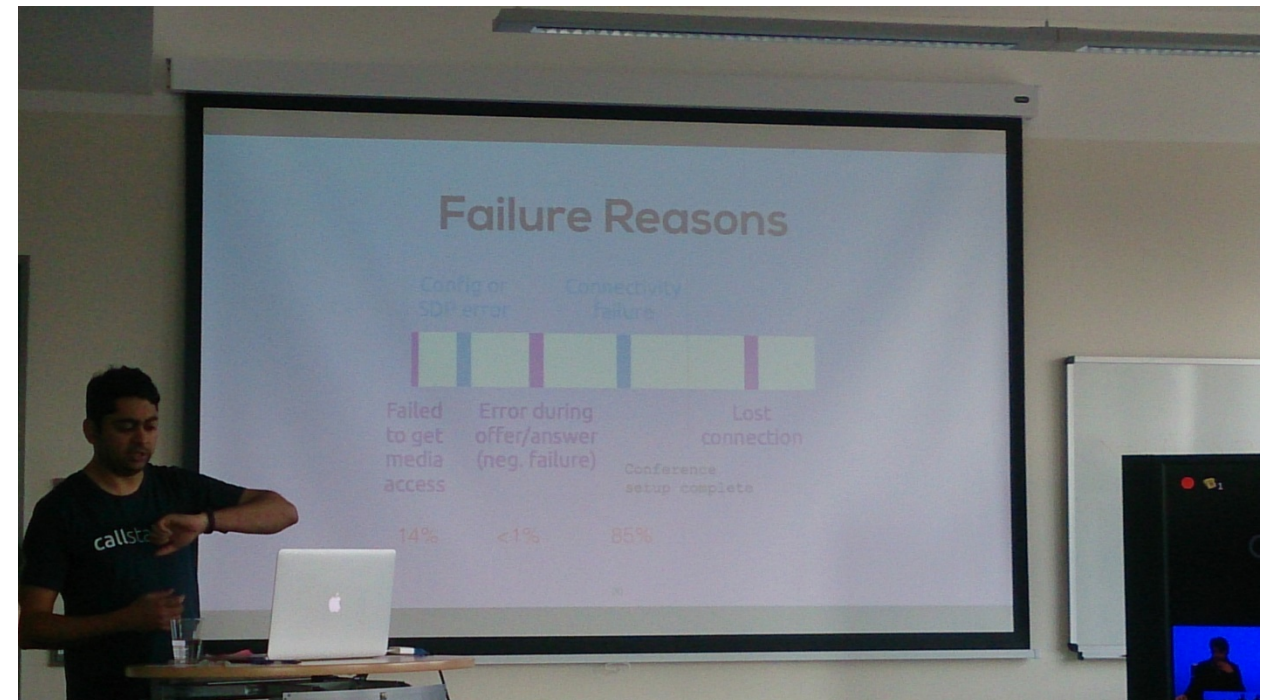
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- When behind NAT or firewall your media connection will fail
- ICE STUN/TURN solves this: mandatory in WebRTC
- Detail but important: if it is not there “WebRTC is broken”
- Do we need a STUN/TURN service and if yes, how can we provide one?
  - Key problem 1: user needs access to a STUN/TURN server
  - Key problem 2: the service specifies which STUN/TURN servers can be used
  - Key problem 3: need authN (ghost SIP calls anyone?)



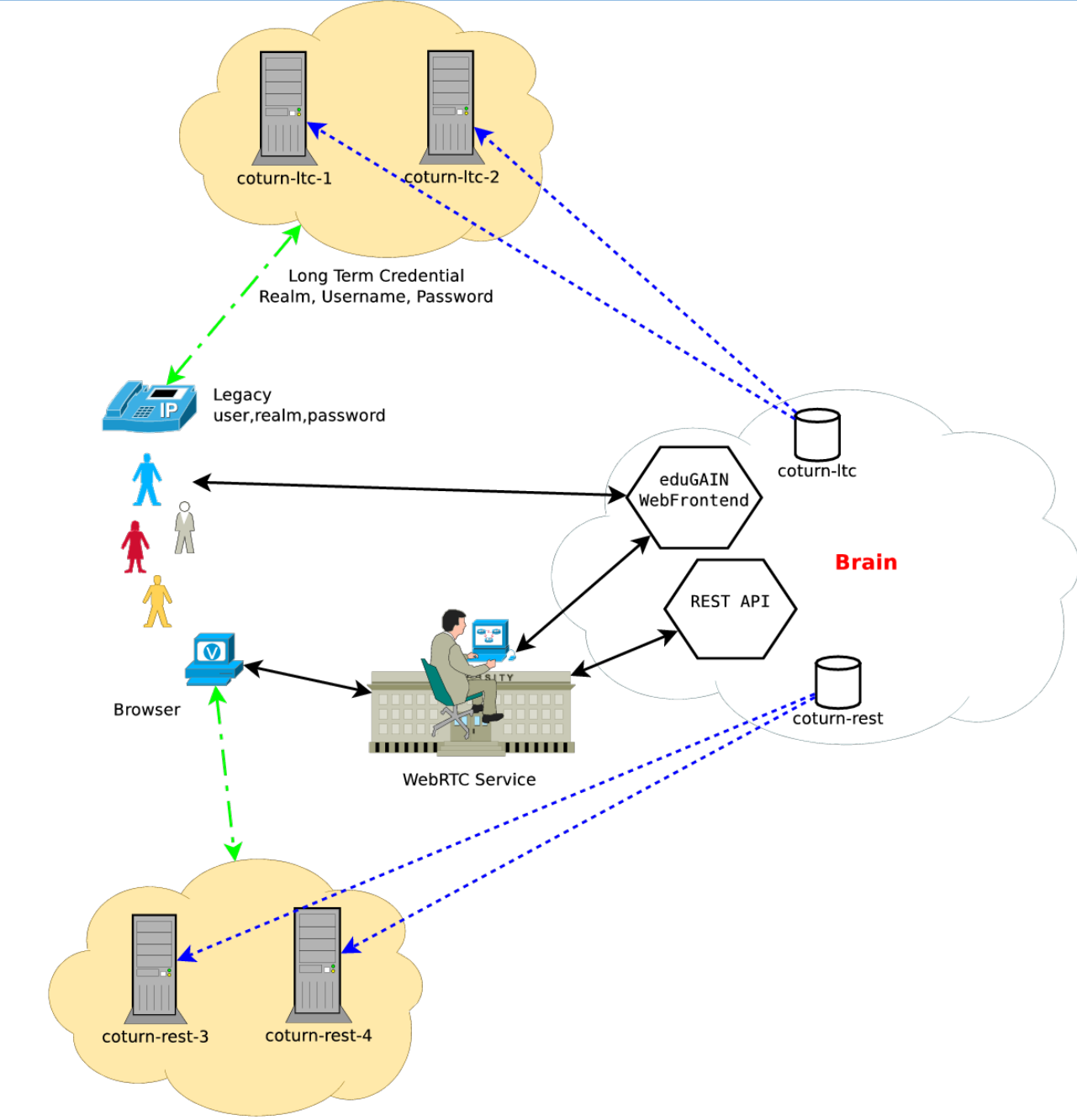
# STUN/TURN Technology Scouting

- STUN/TURN availability is a crucial enabler: without it min. 15% of conversations won't work "WebRTC is broken"
- STUN/TURN infra needed to support:
  - legacy real time communication infra components
  - boxes to support both WebRTC and legacy RTC (Pexip, etc.) that require a stun/turn service
  - developers starting to leverage web-rtc p2p for in-context RTC applications
- Current commercial services have unfavorable business model: pay by GB relayed
- Use of STUN/TURN is only starting





# STUN/TURN Tech Scout PoC



- PoC based on open source coTURN (widely used)
- issue short-lived credentials based on API key gotten via eduGAIN service registration. Services can then just point to STUN/TURN service
- Support for legacy STUN/TURN usage (username/password) as well
- Hands-on experience with STUN/TURN technology and investigate feasibility of a distributed build-your-own infrastructure in EU leveraging NREN networks, IaaS. Possible gains for security.

Shared Secret  
 Time Limited Long Term Credential  
 REST API

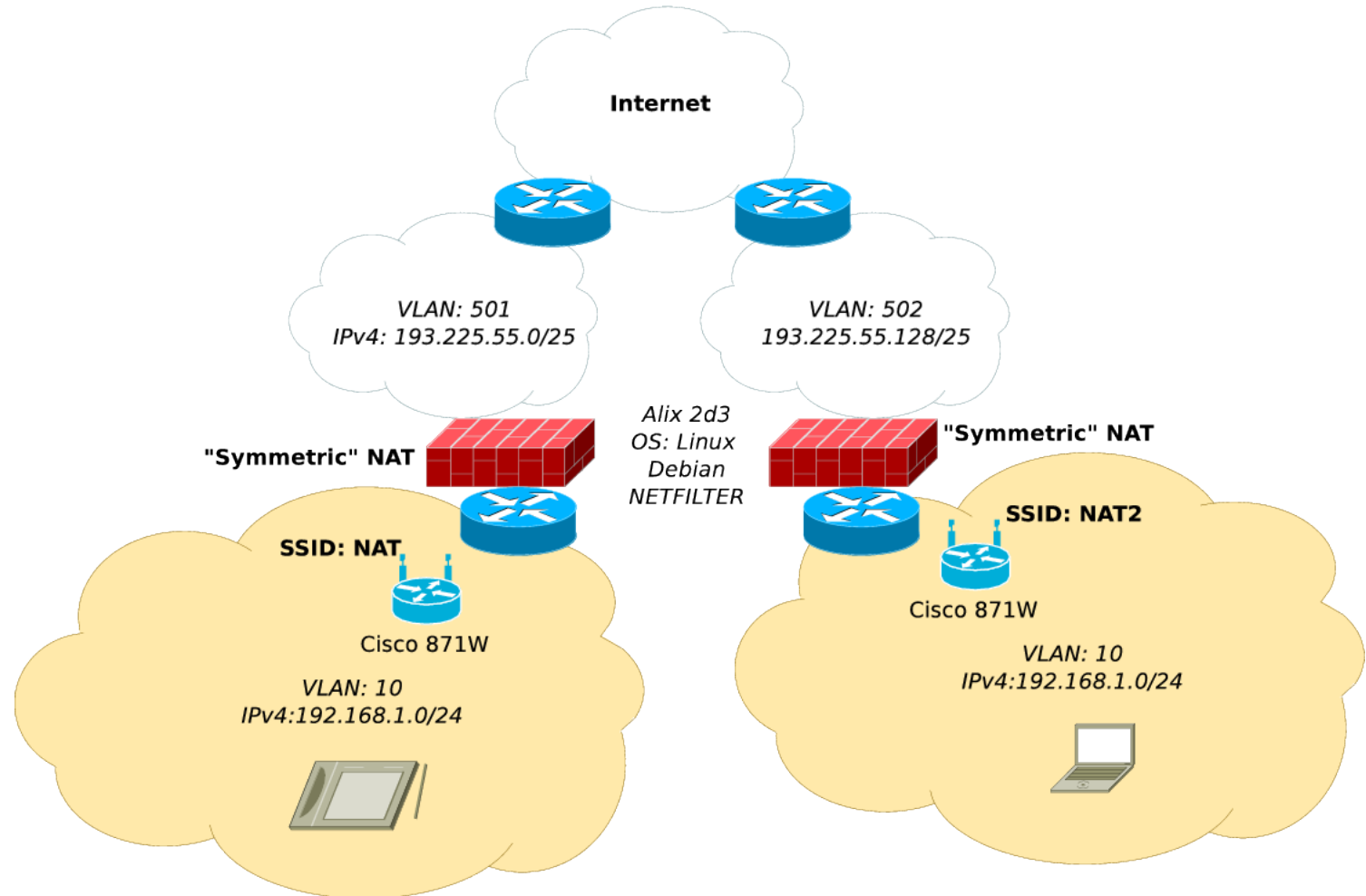
## STUN/TURN recommendation

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- Use of STUN/TURN is only starting
- Productize PoC as part of GÉANT
  - take some years to consolidate demand in EU R&E
  - cheap way to support the three use cases without much risk
  - revisit decision on build-or-buy at the end of GÉANT phase 2

# Eating some STUN/TURN pudding

- Using WebTUT
- Teacher and student @ home
- One demo with, one without STUN/TURN



- Web tutoring service: link expert to non-expert, in this case teacher to student
- Queue + 1-1 real time communication
- Can imagine N-1 or 1-N scenarios, like oral exam, group tutoring
  
- Goal: gain hands-on experience with effort required to do in-context development
  - library quality
  - how much effort goes into real time communication bit, how much in the actual application

Read more about it...



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## GÉANT WebRTC Roadmap + background materials

<https://wiki.geant.org/x/LoJkAw>



Thank you and any questions



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