EENA Technical Committee Document

Public Safety Digital Transformation

The fundamentals of Voice over IP

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1 Executive Summary

Access to Emergency Services for citizens has been through telephone services over the past decades, and through unified emergency numbers like 112 in EU member states and other geographies beyond the EU, 911 e.g. in North America, or other well known national emergency numbers like 999, 000, 100, 101, just to name a few. These services have been implemented into telephony networks according to the available technologies of the time. The last major renovation of phone services that affected emergency calling as well was the introduction of 'ISDN' (Integrated Services Digital Network) with most service providers in the early 1990s.

Since then, telephony was an ever-stable service, working as designed, flawless without major difficulties, but also without major innovations.

With the advent of the Internet in the 1990s and its vast adoption by enterprises of all kinds as well as consumers (and citizens) in the early 2000s, the associated Internet technology, mainly represented by the most popular and widely known 'Internet Protocol' (IP), was a pacemaker for networking technologies which affected all areas of our daily life, at work probably earlier, but in private life and at home perhaps more radical.

Being 'online' and consuming Internet services like the 'World Wide Web' (WWW), 'Social Media' (SM) and 'Messaging' or 'Chat' with a phone (which citizens assume to have an aspect of mobility without any wires and being a smart device) is very common.

Also, the standard telephony services saw the adoption of this new technology, enterprises and service providers were the first to adopt the Internet technology in order to optimize their networks for higher efficiency and lower total cost of ownership in the early 2000s. 'Voice over Internet Protocol' (VoIP) is a technology that was introduced into markets as a successor technology to ISDN and digital telephony.

VoIP as a technology is well proven since the early 2000s and has been deployed in numerous customer environments in all industry verticals, such as financial markets, retail, manufacturing, service providers, in both areas 'Unified Communications' focusing on the communication needs for individuals, as well as Contact Centres at the interface between customers, customer service departments, and service organisations.

Local, regional and national government organisations also started to move to VoIP services over the past few years, mostly with the objective to benefit in the area of Unified Communications, offering benefits to workers and contributors, as well as IT and operations departments.
In some countries also public safety and emergency services did adopt VoIP already a few years ago, but still being seen as early adopters in this specific industry vertical.

Over the past years since the introduction and adoption of VoIP the reception of this technology has moved from being seen as an ISDN-replacement-technology to rather an enabler technology for digital transformation, as it very much aligns with Internet-oriented applications and data networking technologies, supporting and enabling current "4.0-trends" like "Industry 4.0", "Healthcare 4.0", "Government 4.0", all representing their specific flavour of digital transformation in a specific industry vertical.

With a closer look into public safety, VoIP offers the potential to support overcoming some of the imminent challenges and clearing the path to modernized emergency services for all citizens, leading to "Emergency Engagement 4.0", also discussed under the label "Next Generation Emergency Services", and under a technological aspect driven as "Next Generation 112" by EENA, ETSI and other organisations and bodies.

This document is aiming at outlining the benefits of 'Voice over IP' to operational people in public safety and emergency services in the context of current efforts from service providers to sunset their traditional phone services with ISDN and moving fully to IP-based technologies under the label 'All-IP' throughout every region of the world.

The document introduces some basics in VoIP technology understanding, pointing out why it is relevant to the emergency services business, describing necessary pre-requisites, looking into strategic developments linked to the end-to-end presence of VoIP, and closing with the traditional EENA recommendations to the different stakeholders involved.

2 Technology Explained

In order to understand the differences in traditional voice communication technologies and 'Voice over IP' as well as 'SIP', it sometimes is helpful to get an understanding how these technologies work, allowing a more informed decision making on the path of modernisation.

The appendix in this document outlines the basic technologies that are used to build telephone networks, with the objective to give interested non-technical readers an insight in which way these technologies are enabling voice communications and how they have been developing over time.

In principle the concepts of 'Circuit Switching', representing the traditional technologies in telephony, and 'Packet Switching', being the basis for 'Voice over IP' will be explained in order to allow building an understand of how transport of a voice connection over a data network works and what the challenges embedded in this approach are.

Also in the appendix there is an overview describing a more infrastructure oriented view on 'Voice over IP', with a look at Local Area Networks (LANs) and the capabilities they need to provide as well as the necessary network engineering guidelines to enable high quality voice services to be delivered.

3 The relevance of VoIP for Public Safety and Emergency Services

This section is going to outline why VoIP can be relevant for Public Safety and Emergency Services, explaining why other verticals have picked it up initially from a commercially driven aspect of network consolidation.

After that, the specific value proposition for Public Safety and Emergency Services is going to be reflected.

3.1 Initial drivers to introduce VoIP

Over time, as IP-based data networks became more and more ubiquitous in companies and the available bandwidth in Local Area Networks (LAN) as well as in the Wide Area Networks (WAN) connecting different sites grew, the appetite of running voice services on these data networks became bigger and bigger.

There are a couple of reasons that can be quite compelling from an organisation's point of view to introduce VoIP in the core of their business, most of them driven from a financial and economic perspective.

3.1.1 Network Consolidation

Operating two different networks, voice and data, became inefficient and expensive. Two different cabling infrastructures are required, a 2-wire phone network and a 4-wire Ethernet network. The biggest financial effort here is the parallel deployment of two different cables to every workspace, one to connect to the phone, and one to connect to the PC.

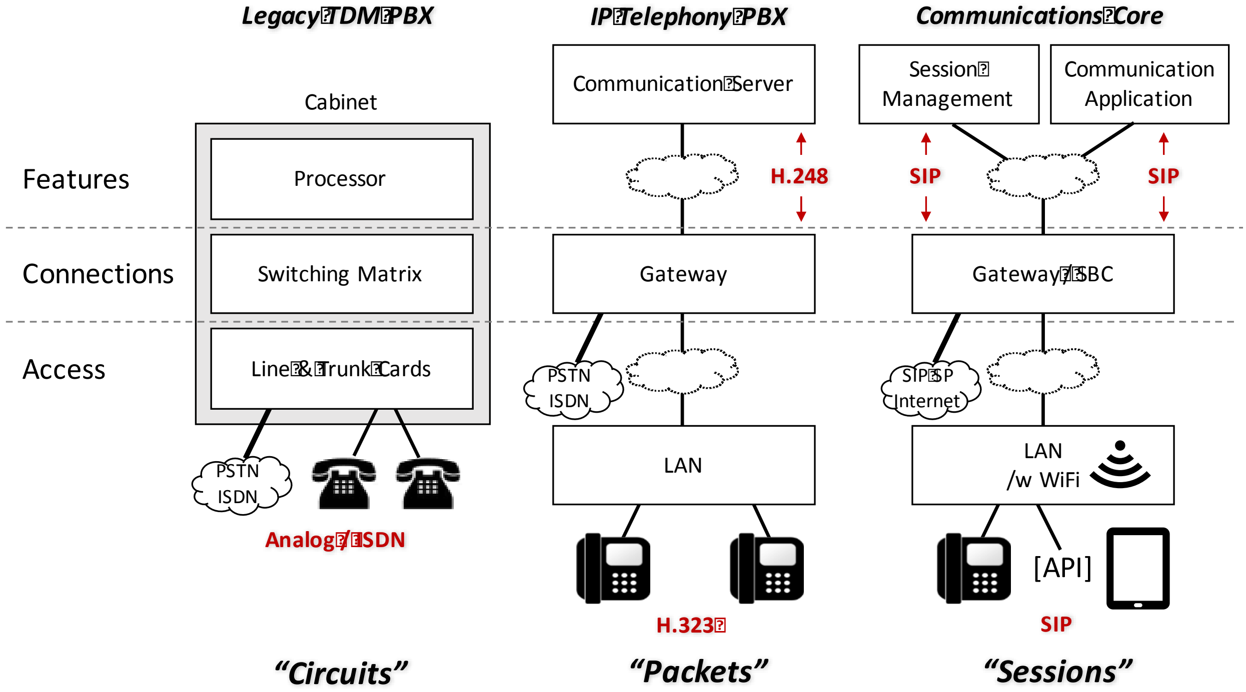
As a consequence, also maintenance for two networks adds to the total cost of ownership (TCO), and therefore consolidation from two networks into one promises to pay off in a reduced TCO.

3.1.2 Distributable PBX Architectures

With 'cutting' the phones from the line cards of the legacy TDM PBX and hooking it up the LAN, there immediately was a way to start building distributed architecture.

The PBX being an Enterprise Telephony Systems mostly has been a single site construct, a monolithic block: its features implemented as vendor-specific software code on proprietary processor cards, connections being controlled by that processor and established in a hard-wired (analog) or software-based (ISDN) switching matrix, giving access to phones and the public network through again proprietary line and trunk cards, all wrapped by a tin cabinet, as presented in the graphic below.

Of course, most of the traditional PBX vendors did have the capability to set up a distributed system across multiple sites, but this came with quite some effort for remote processors, remote cabinets or shelves, all connected to the main PBX through ISDN tie trunks, which were basically dedicated to this single use and were quite costly.



With the advent of the IP-based PBX, most vendors moved their PBX software from proprietary processor cards to more generic communication servers. Connections were no longer set up in the switching matrix, but moved into the functions of the gateways in order to be connected to the outside world using ISDN links to the PSTN (Public Switched Telephony Network).

The move to more industry standard components within the architecture, paired with the introduction of IP-based protocols like H.323 for the IP telephones and H.248 for the gateways allowed the overall architecture to become more distributable, moving away from single side-deployments and being "monolithic blocks" to server-gateway distributable system, allowing a single PBX to span multiple sites, cities, regions, countries. So a "corporate network" became much more manageable, as in fact a single system could drive voice services for the organisation where previously dozens or even hundreds of single-site PBX were needed.

3.1.3 Voice Services Transformation in a SIP-centric Deployment

The evolution of voice services over the past decade brought an incredible level of openness and flexibility to the market, that now can be picked up especially by public safety and emergency services organisations in order to strategically create modernized approaches to 112 and any other emergency number.

Within the first years of the 2000s, SIP as a protocol became more mature and was adopted by many PBX vendors. As mentioned previously, the protocol's characteristics are much closer to web applications than to traditional voice services.

And with SIP being an IP-based protocol that also now is applied to the connection between the enterprise world and the service providers public networks, also the last domain of vendor-specific hardware was collapsing: special telephony ISDN interface cards were replaced by standard Ethernet connections between routers and switches. In case a dedicated service demarcation point for the voice service is required between the enterprise and the public network – in analogy to the corporate firewall protecting the corporate application platforms against attacks from the Internet – this capability is provided by a Session Border Controller (SBC). And SBCs can completely be built in software on standard industry server hardware.

With SIP, sometimes even telephones built in hardware are no longer involved in a voice communications. Although many organisations still prefer a "real" telephone on the user's desk, many have bought in already into the idea of making every possible device a telephone, starting with the obvious and already existing approach of a softphone on a stationary or mobile PC platform. With the fast and huge adoption of Smart Phones and Tablets in the consumer market from 2007[1] onwards, these platforms became very interesting for

[1] Apple introduced the world's first real smart phone in 2007, changing the way that people communicate radically.

enterprises as well, so currently[2] every leading vendor of PBXs offers an app-based soft client to work with Android, iOS and other smart device operating systems as well in order to extend the reach of corporate voice communications beyond the physical "borders" of the organisation.

That being said, we currently see the latest interpretation and step forward: Voice Services being delivered as an API (Application Programming Interface) only. Why is that interesting? Because in principle this allows third-party-organisation or even customers to build their own communication applications, in exactly the way that they want, maybe with a special vertical application focus in mind, consuming the multitude of communication services (where voice is only one of them) through the optimally shaped, purpose-built and bespoke developed device. Just think about your next call taking or CAD application and the communication capabilities you would want to have built right into it!

In fact, all of this made organisations change the way that they look at how they want to leverage voice communication for the purpose of their business, they become very similar to other application services:

- Centrally hosted in data centres
- Deployed on virtual machine infrastructures for increased efficiency and availability
- Consumed in different enterprise locations or even outside of the enterprise

Just to add on top to that, many features and functions of the traditional PBX have move into applications managing the SIP sessions (Session Managers), and vendors are becoming more open to give access to these platforms for new communication services to implement new features according to very specific requirements that previously would not have had any chance to be realized because they were too specialized to become an item of any vendor's telephony feature roadmap.

3.1.4 Technology Refresh? Do PSAPs need this?

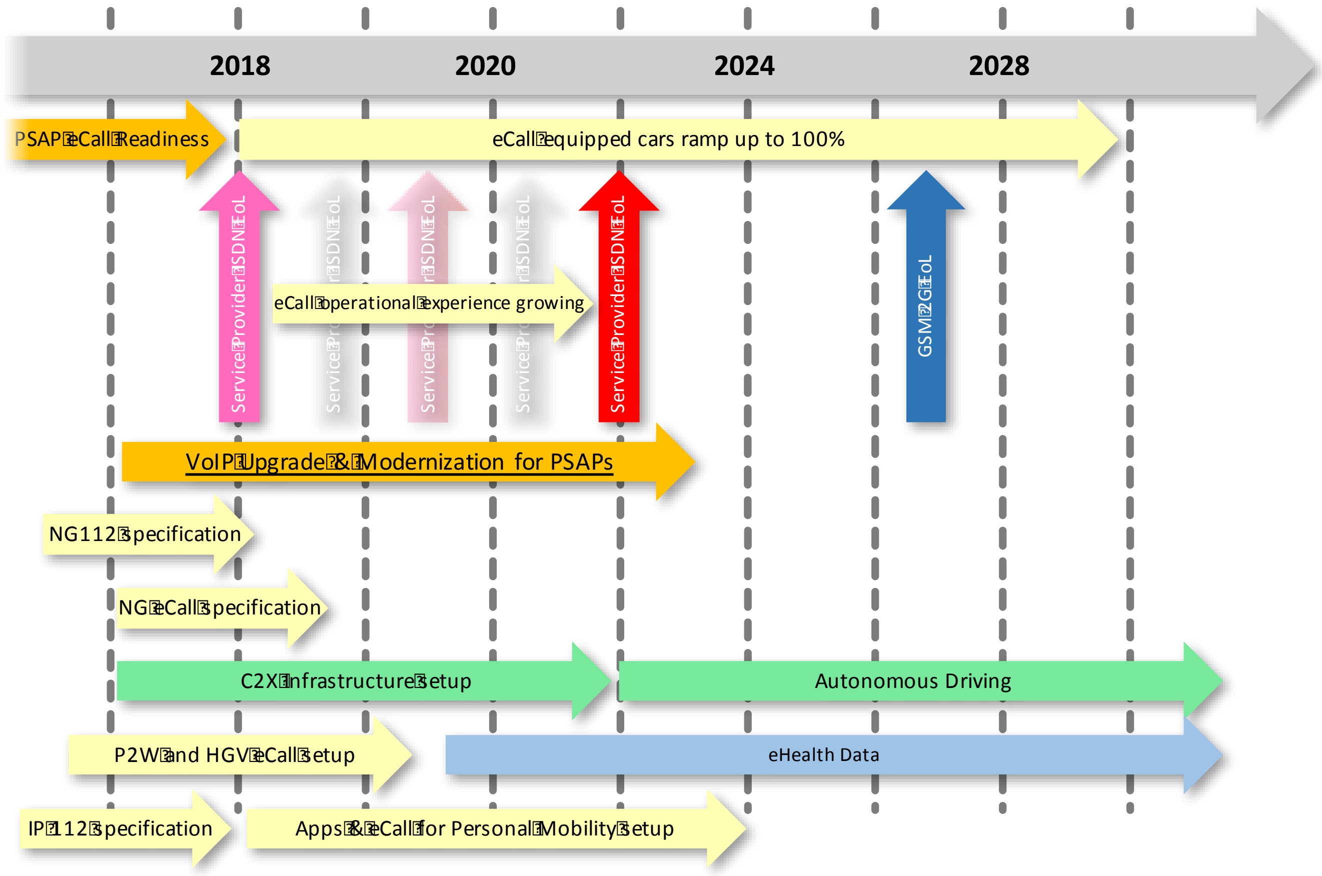
3.1.4.1 Technology Perspective

Watching the changing scene from a technological perspective, the necessary move from ISDN- to Voice over IP-connectivity to the PSAP as well as the timely coincidence of the introduction of EU eCall could potentially be seen as the biggest drivers for a conversation on technology refresh.

The compelling event for this technology shift can be seen in the service provider networks that have moved to IP-core networks by deploying an Integrated Multimedia Subsystem (IMS) – moving to VoIP internally – for managing the multimedia real time traffic on the same network infrastructure that was introduced for data (a.k.a. Internet) traffic. Now, in a last step towards a full migration to IP, service providers start to replace the last mile that today connects enterprise customers as well as PSAPs using ISDN trunk technology with modern SIP trunks.

But this effect, as well as the introduction of eCall, should be seen in the wider context of what we expect to see happening over the next years, as shown in the following graphic:

[2] February 2017



From an operational perspective of course the introduction of the EU eCall is a big driver for change, as eCall is the first new service mandatory to be implemented within the EU. Although technologically not directly related to VoIP[3], EU eCall it is related to the transition to VoIP, as it appears in a timely coincidence[4] of many European Service Providers discontinuing their ISDN access services, e.g. as announced by Deutsche Telekom for 2018 and Vodafone in Germany for 2022, just to name a few. Other service providers have announced the 'End of Life' either publicly already, or are at least planning for it internally, with an expected timeline for up to 2022.

This is of course affecting all PSAPs in the specific countries if they have not yet moved to SIP[5]. But yet, moving from ISDN and introducing eCall in the same period of time is considered the tip of the iceberg regarding drivers for change.

A serious change in scope of emergency calling is expected to come from current standardization initiatives to be completed within the next 1-2 years, thus expected to be picked up by European and national regulation:

- **"IP 112"**
  touching the immediate transition to SIP-based access for PSAPs as an intermediate step towards NG112
- **NG112**
  focus on new principles of emergency call routing and embedding additional data/location information into the call to 112, assuming End-to-End SIP-connectivity in the emergency calling chain

[3] EU eCall technically speaking is a standard voice call using an in-band modem to transmit additional data from the car involved in an accident to the PSAP
[4] Due to the delay of the introduction of EU eCall from initial plans of 2013 to finally 2017/2018
[5] As of 2017, we assume that >90% of all PSAPs in Europe are connected to the public network using ISDN lines

- **NG eCall**
  moving away from in-band modem technology to 4G/5G originated eCalls based on SIP and transported by SP's IMS networks, with new capabilities in terms of 'connected car' for PSAPs to manage incidents

Operational experience with eCall and potential new eCall-related services between 2018 and 2025 might lead to a requirement to adjust technologies, services and procedures:

- **Managing synchronicity and coherent eCalls**
  With the adoption of eCall in new cars, the real development of eCall-related 112 traffic will become visible. Especially the amount of calls and the respective management of these calls will have to be evaluated, as it might occur that incidents like mass accidents can result in short peaks of incoming automatic as well as manual eCalls, augmented by the regular 112 calls.

- **Extending eCall scope to P2W and HGV**
  There are ongoing conversations on adding eCall capabilities to powered 2-wheelers (P2W) and heavy goods vehicles (HGV), which could result in a higher number of potential eCall originators, as these new services could be agreed on and added to the eCall services portfolio. Due to the different parameters and requirements between passenger cars, heavy goods transports and motorcycles, this potentially leads to changes how to treat these calls from a procedural perspective.
  Probably there will be no voice call element to a motorcyclist that went off his bike, but first responders need to be prepared to search for the rider in a different place to where the bike originated the call.
  Also with HGV, the eCall MSD will contain data on the loaded goods or references to this data, so PSAPs will need to be prepared to take this into account as well from a procedural perspective.

Personal mobility and usage of emergency calling apps will add additional context to 112 calls

- **Smartphone Apps**
  Smartphone apps have the potential to help solving the current challenge to convey the exact location of the caller to the PSAP in addition or as an alternative to network-provided mechanisms implemented by the Mobile Operator or Advanced Mobile Location (AML) implemented with support from Google.

- **Smartphone apps and eCall Personal Mobility**
  EU eCall with its in-band modem offers a well defined approach to transmit caller location data into the 112 PSAPs. There have been considerations if that kind of technical approach could be picked up for mobile (smart)phones as well, helping to overcome the immanent challenge of knowing the precise caller location.

New technologies and concepts with potential influence on emergency services are appearing in the conversation, and we currently do not clearly see what kind of changes they could be bringing to emergency calling:

- **C2X Infrastructure**
  Car-to-Everything Infrastructure is taking into account that cars will be able to communicate amongst each other, and also to roadside infrastructure, exchanging information on traffic and critical situations ahead.

- **Autonomous Driving**
  Are we going to experience less accidents when all cars will be machine-driven and the human element of danger will be excluded? Will eCall still be relevant then? On the other hand we also expect that the transition period for 100% autonomous driving which could indeed reduce the risk of traffic accidents dramatically will not happen over night, but rather counted in decades.

  This means that the human factor will be around for quite some time, and the co-habitation of human and autonomous driving may lead to unforeseen effects in terms of traffic-related emergencies.

- **eHealth Data**
  In the EU there is an ongoing debate on eHealth and the electronic patient record. As this conversation is expected to mature over the next years, there seems to be considerable value in

adding patient-related data to emergency calls already with the call. SIP-based networks are going to help getting this accomplished, and in the end we expect the PSAPs being able to leverage this data for more accurate emergency response, but also being more connected in the overall emergency response and patient care chain, with again side effects on implemented procedures.

3.1.4.2 Operational Perspective

Under the question "Who needs technology refresh in the PSAP?" the aspect of managing overwhelming mass calling scenarios related to natural or man-made disasters can be addressed with intelligent solutions, taking into account distribution of calls to other PSAPs to overcome overload scenarios.

This of course leads to a requirement for enhanced collaboration between different PSAPs and probably between different agencies, to be accomplished by connected PSAPs.

Another challenge in many countries is associated with a high amount of false calls and/or nuisance calls, which have the potential to affect the overall performance of the 112 service. Also the appearance of Denial of Service Attacks has been noticed during the past years, affecting the service availability.

Understanding these service-affecting patterns and creating solutions like blacklists and whitelists is a possible approach to reduce the operational risk.

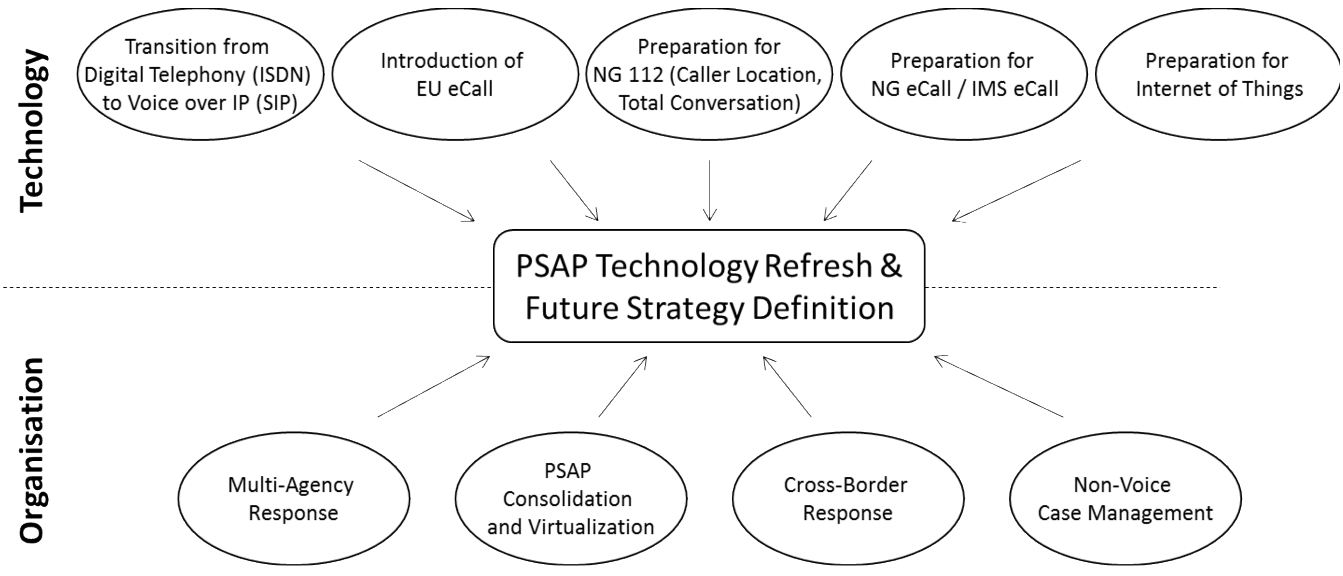
Beyond that and with regard to the other topics mentioned, there is a need for higher technological flexibility than we experience it to day, and a high level of openness in the PSAP platforms in order to be capable to integrate new technologies as they appear.

3.2 The Value Proposition of VoIP in Public Safety: Overcoming Current Challenges

3.2.1 Challenges for emergency services

In context of the aspects mentioned in the previous "Technology Refresh" discussion, a wider value proposition of VoIP as an instrument to adopt to new or changed requirements is already established in the public safety vertical:

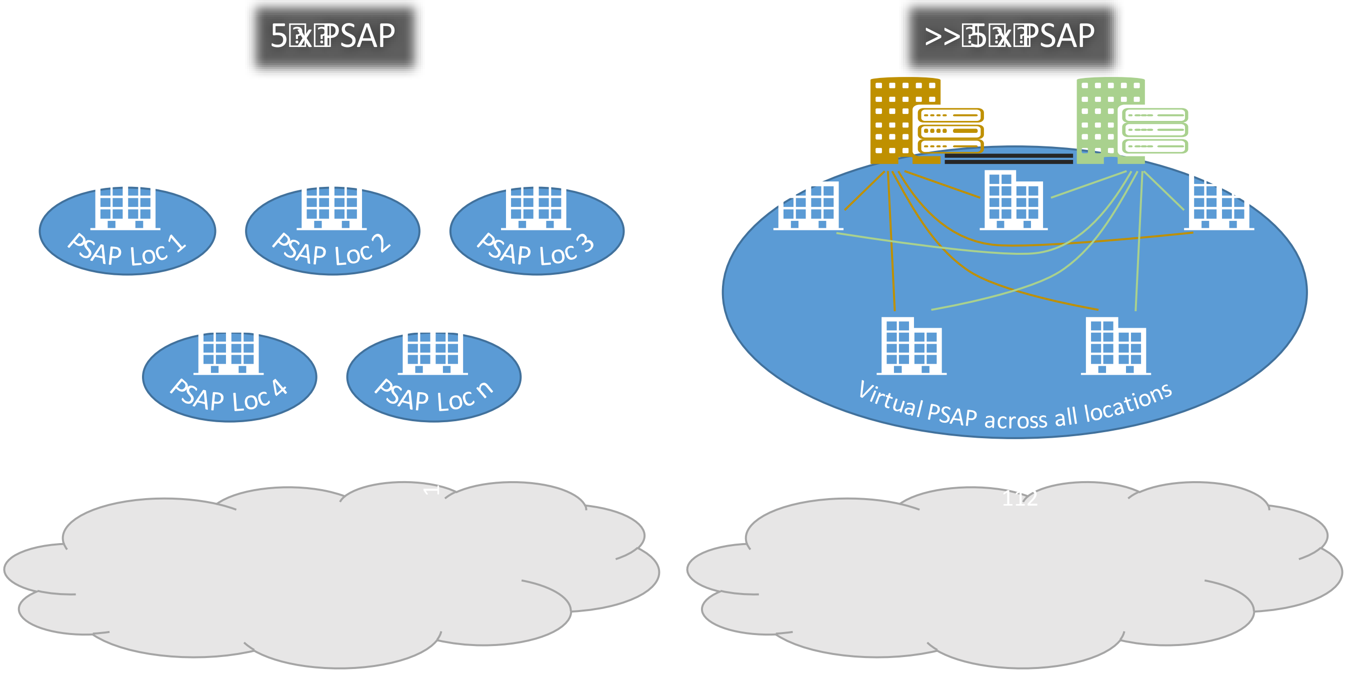
- Preparing for Next Generation 112 as mentioned before builds on the presence of SIP-based communication end-to-end in the end-state. On top of that, new access channels as evaluated in REACH112 ("Total Conversation") and additional data like caller location information embedded into the incoming contact can easily be created.
- Preparing for NG eCall, whilst still introducing EU eCall, is a side effect of moving to VoIP in the PSAP.
- Evaluating and preparing for the Internet of Things can be seen in the same technological context, and should initially leverage progress being made in preparing for the previous points above.



From an organisational perspective as well as from a procurement point of view, all of these new technologies can be seen in conjunction with each other. Beyond understanding them technically, the discussion of the operational perspective is likely to unveil synergies here:

- Multi-agency and cross-border response capabilities can be built on new and open platforms, without neglecting different service requirements, views and legacies.
- Consolidating small PSAPs into more flexible and larger constructs, offering better capabilities of handling larger incidents from a capacity perspective can immediately be achieved by linking into an updated core.
- Managing cases with other channels than pure voice calls coming in will be experienced when preparing for EU eCall services to become operational from October 2017 onwards. In fact, even if EU eCall is still a voice-only call, PSAPs are going to open up a data channel which needs to be adopted also from an operational perspective, especially reviewing existing and well-established procedures[6].

With regard to the consolidation of multiple smaller PSAPs into larger constructs, let's have a look at traffic theory here. Without going into too much detail here, it is a fact based on statistical approaches known as Erlang's[7] traffic theory that e.g. 5 PSAPs with 6 call takers each can carry less traffic than the same 5x6=30 call takers in a single bigger PSAP:



For a deeper insight into the principles of traffic calculation there is a specific section in the appendix of this document.

On the other hand, not for every country's governmental structure allows to implement such an approach without the need to change country laws, which of course very often may not be seen as an option.

3.2.2 Aspects of PSAP consolidation and virtual large PSAPs

Whilst in reality the devil is in the detail and a PSAP's efficiency would not be judged under the aspect of a theoretic approach (see appendix "Traffic Calculation"), we can take away that driving collaboration between call taker communities and mowing towards a construct of connected PSAPs or even virtualized and distributed PSAPs is raising the organisation's capability to better manage overwhelming call events.

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[6] On a personal note from the author of this document, EU eCall has got more in common with the IoT (sensors, additional data, process automation) than with a standard 112 emergency call: "eCall is the introduction of the IoT into the PSAP world in disguise of a voice call."

[7] Erlang Description, Erlang B and Erlang C

Of course we would need also to consider that simply distributing call events that cannot be managed by one PSAP to all other PSAPs in the area of responsibility could lead to block all of those PSAPs in the case that still the incident's amount of calls would be overwhelming even to the much bigger community.

In fact simply connecting PSAPs or creating a national virtual PSAP alone is not everything, we would to also investigate to create a set of rules to distribute calls and not flooding all PSAPs at the same time. These rules of course would need to work on data associated with the incoming calls.

In today's world, the only data available immediately with the incoming call is the calling party number and very often "Cell ID" from the Mobile Network Operator, which is used to derive location information from it, either crawling the service providers address database in case of fixed line emergency calls, or accessing the mobile network operators cell database to determine the geographic area of call origination.

The next chapter is looking into different kind of "additions" to the current as-is situation.

3.3 VoIP as an "enabler" for advanced Emergency Communications

As discussed in the previous chapters, VoIP and especially SIP are seen as enabler to more advanced communication services, due to the protocol's capability to carry more information with the call than just the calling party number.

3.3.1 Scalability and "Elasticity"

With the current way of sending 112 emergency call traffic from the service provider's networks to the PSAP based on ISDN trunks, flexibility to dynamically adopt to overload situations is very limited, if not impossible at all in terms of 'time to action'. ISDN access technically is bound to hardware boards installed in the PBX, with the capability of running two simultaneous calls over a 'Basic Rate Interface' (BRI), or 30 concurrent calls over a 'Primary Rate Interface' (PRI).

In most traditional PSAP installations we find a number of BRI or PRI interface cards being installed on the PBX. This is then exactly determining the number of concurrent voice channels, which typically is a somewhat higher than the number of maximum staffed call takers. Within smaller PSAP environments with 2-4 call taker seats, we very often find 4-8 voice channels (equalling 2-4 BRI interfaces) available to take emergency calls. In larger environment, we very often find 2 PRI cards, resulting in a capacity of 60 simultaneous calls.

When it comes to a major incident and an overwhelming amount of 112 calls, the amount of traffic firsts exceeds the available call takers, resulting in a degradation of service levels (wait times can grow from a few seconds to a number of minutes), and then - once the number of available voice channels is exceeded – blocking additional incoming traffic.

In case the capacity for incoming calls would need to be expanded, the following activities need to happen:

- PBX in the PSAP would need to be equipped with additional BRI or PRI cards
- More ISDN lines need to be ordered with the Service Provider
- Configuration and Installation typically require a lead time of 2-6 weeks before capacity extension is completed

Looking into a SIP trunk-based approach, additional capacity very often should be nothing more than a request to the SIP service provider, followed by a commercial agreement, to add more capacity to the SIP trunking service.

Remember that a SIP trunk technically uses an Ethernet connection (see Appendix chapter A.5), capable of providing a bandwidth of 100 Mbit/s or more, and thus carrying as much as hundreds of simultaneous voice channels. So the flexibility in terms of access capacity is huge with SIP trunks.

That being said, we could event think about a fully automated process (think Digital Transformation!), monitoring current traffic load, and requesting additional SIP trunk capacity with the service provider when needed.

And to add to this, we could even think about creating additional 'virtual call takers' in terms of speech-enabled automated applications which only kick in under the condition of overload and highly exceeded call taking capacity, helping to qualify and filter the incoming calls before involving a real call taker resource.

3.3.2 More context: location, patient data, pictures

Probably the most interesting and most relevant context to an emergency call is precise information on the location where the incident happened and where emergency response staff needs to go. This information in most cases currently is inaccurate (e.g. Cell information from the mobile network operator, pointing to an area of square kilometers), not available (e.g. no cell information), outdated (e.g. billing address from a service provider's database), or not reflecting reality (e.g. billing address from a service providers database for an enterprise that running a cross-regional corporate network, connected to the service provider through a SIP trunk at only one location, reflected by the billing address).

Managing access to more precise location information with the incoming call is probably very important in the future.

Another type of context could be patient data attached, a picture from the incident shot by the smartphone that originated the call, just to name a few.

Additional context creates additional insight, leading to better informed decision making and overall better emergency services quality, for both citizens as well as the emergency response organisation.

3.3.3 More channels: Text, SMS, Social Media, Video, Total Conversation

Smart phones are around for about 10 years, and citizens have fully embraced them with all of their capabilities. And they have changed the way that we communicate as well as the culture of communication. Whilst a solid conversation on the phone is a cultural heritage that many of us born in the 20th century still are able to intuitively work with, especially the generation of millennials started to endeavour visual or textual communication as their preferred way to get in touch with others.

That being said, many people are not that much used to making phone calls any more, and therefor in a situation of danger, irritation, confusion, disturbance or panic would probably be not as much prepared for a decent voice conversation, so would prefer texting or even using pictures or video to communicate, just like they are used to do this in their peer groups on a daily basis.

Deriving from that, the question of what media to integrate is widely discussed. The most obvious pone probably is the SMS service, as it is basically known and understood by everyone. From an emergency service perspective only few countries really have introduced SMS messages to 112 as of now.

Beyond SMS texting any are of the widely accepted social media services like Facebook and Twitter come to mind immediately. Watching the conversations and discussion around this, at least monitoring twitter and reaching out to the general public through this service became a solid instrument to manage large scale events like natural or man-made disasters, where voice services simply collapse due to overload[8].

Especially amongst the younger citizens messaging services using Instant Messaging has become very popular and is currently surpassing the use of "classic" social media in terms of active users on platforms like e.g. WhatsApp and Facebook Messenger.

With this kind of services a method of centrally manging response to requests on these channels is becoming increasingly more popular: chat bots, using the mechanisms of Artificial Intelligence to constantly grow their repertoire of automated conversations. We can easily imagine how this technology can be applied to public safety in order to understand the context of a request, pre-triage the incident, and this making an informed selection of who should ultimately answer the request, or if it should be answered at all under the impression of high volume traffic.

Also, it would be quite easy to apply online translation services to such kind of a text-driven conversation, expanding the multi-language capabilities in a more and more multinational society, covering more languages than actually been spoken by the call taker staff.

Another aspect could be to expand the 112 "brand" to websites that can be access by any device with internet access, even without the capability to make a call because they don't have a SIM card ("WiFi hotspot surfing"). Surfing to e.g. www.112.de could be an interesting approach to open up communication channels, that could easily use e.g. smartphone or tablet location data to route the session to the most appropriate PSAP.

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[8] refer to Bataclan video

With that website-approach we can also see that leveraging WebRTC[9] may lead to a new form of voice and video communication from basically any device connected to the internet.

So we expect a lot of additional value also for people for whom a call to 112 simply is not an option, due to restriction in their capabilities to talk or listen, or to the situation that they are in:

- Citizens with difficulties in hearing or talking, e.g. the community of deaf people
- Citizens with insufficient language capabilities, e.g. tourists or refugees
- Emergency requests from noisy environments
- Emergency requests in situations where talking can add an additional element of danger, e.g. burglar in the house

In a special consideration, adding video to the context would help especially deaf people to use sign language interpretation as an appropriate means to communicate fast and efficient with emergency services.

In summary, many of these multi-channel capabilities reside within the capabilities of IP-based PBX system and their feature extension, and they are well proven already in the world of commercial contact centers.

3.3.4 Better Collaboration

One of the challenges today is the limitation of modern ways of collaboration between different organisations involved in managing an incident or even a crisis situation. Exchanging data still very often is limited to voice calls or sending a fax from PSAP to PSAP[10], sending data immediately with the call to allow for tighter process integration is not widely discussed as of now.

As introduced earlier, moving to SIP would easily allow organisations to create standardized data exchange formats to transmit case-related data immediately with the call, just as natural as an attachment to an email. And in terms of integration into processed and procedures, this data could be elegantly picked up from modern web-based CAD and GIS applications, or it could even reside on a shared database, allowing to send references to a case's data and not the data itself in order to avoid aged data and data inconsistencies.

Another element not very widely used in emergency-related real time communication is conferencing. For instance the upcoming integration requests from third party services in eCall (TPS eCall) could be an excellent use case for conferencing, as they have been taking the initial emergency request and could be of value to stay in the conversation when handing over the session to emergency services.

Conferencing in fact could be a new concept in emergency response:

- Voice conferencing can be used to add specialists of any kind to a conversation, e.g. psychologists in a threatening situation, staff trained in epidemic crisis management, medical experts, etc.
- Conferencing could not be limited to voice, but also include web and video collaboration for a richer communication experience, e.g. sign language interpretation
- Conferencing could be a very appropriate way for inter-agency collaboration

Conferencing solutions including voice, video and web collaboration today are part of PBX vendors' portfolios, so again VoIP can be an enabler for these services to extend the citizens' experience beyond a simple voice call if need.

3.3.5 Managing Security and Denial of Service

For a very long time voice networks have been seen as very secure with regard to denial of service attacks. With ISDN networks being somewhat specialized, also due to the high level of "technicality" of the telephony oriented protocol, quite a high effort (so-called auto-diallers) was needed to misuse the voice network intentionally and undetected.

Having discussed the benefits of VoIP and the rationale to move to SIP-based communication, its lightness and its proximity to open applications, this also turns out to be a flip side of the coin: like cyber attacks have been occurring with the advent of the Internet immediately, we also see the same phenomenon arise with communication services. Intrusion into networks, communication hijacking and other misuse of communication services with the objective to degrade communication service (Telephony Denial of Service,

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[9] Web Real Time Communication, a method to run voice and video from within a standard browser
[10] On a personal note from the author: „Fax" and "Emergency" is seen to be a contradiction in itself in 2017 ;-)

TDoS) or commercially misuse the services in an unauthorized way (toll fraud) are a reality and lead to quite some amount of damage.
To be clear here: TDoS attacks exist with and without IP telephony, the difference is that on the one hand a large number of small SIP service providers are in the market, and some of them have not secured their services against abuse in the most professional way, opening the door to abuse their services without any authentication of the calling party, or even with false caller IDs.

On the other hand, the tight linkage between web applications and telephony services in consumer's best friend, the smartphone, creates an environment that can intentionally or unintentionally be leveraged for Distributed Denial of Service (DDoS) attacks, resulting in a huge amount of calls.

How does this impact Emergency Services today, without having moved from ISDN to SIP? There are number of scenarios that typically can occur, even in combination:

- Overload of calls
- IP network unavailability
- PSAP unavailability

Within the last few years also emergency services have been attacked through unauthorized use of SIP services, basically following the same patterns:

- VoIP on the attacker's side facilitates abusive origination of large numbers of calls, inexpensive and easy to automate and to repeat (occupying emergency lines)
- A simple programme triggers phones and tablets to continuously call 112/911 (few lines of code and a tweet would suffice)
- Botnets making thousands of malware-infected devices launch automated DDoS attacks to cripple emergency phone systems

Protecting against these attacks is not practical in a classic 112 environment today, because all calls to 112 must immediately be routed to emergency services, regardless of the caller's identity.

Although potentially a wider conversation on preventive measures is necessary[11] based on the PSAPs and Emergency Response Organisation's risk profile, moving the telephony access for the PSAP from ISDN to SIP does offer a compelling advantage: modern Session Border Controllers needed as a demarcation point to separate the PSAP's network from Service Provider's network do offer security features that can be applied to help solving the overload situation of TDoS and DDoS attacks.

Any type of DoS attack that is directed against the PSAP's SIP trunk or one or more phone extensions in the PSAP does come with a specific attack pattern of SIP messages that can be identified by an SBC. Based on these attack pattern, different deployment thresholds for SIP messages shall be configurable in the SBC. These thresholds are global. Avaya SBCE enforces these thresholds based on the source of an attack.

Looking at the SIP trunk level, deep inspection of SIP messages will allow blocking of incoming call request, e.g. 20 SIP messages per 5sec coming from a single source can be an adequate configuration.

But also other type of DoS attacks can be detected by an SBC, and counter measures be taken by intelligent configuration of the SBC:

- Stealth DoS or Distributed DoS**,** where the source of the attack call is constantly changed
- Call Walking, a type of DoS attack whereby serial calls originating from a single source (normally spoofed) are directed against a sequential group of endpoints (phone extensions in a row, e.g. -251, -252, -253 …)
- Server DoS, attacks against the service infrastructure such as SIP servers

To conlude the security and safety conversation here, we have to accept that attackers have moved to VoIP and SIP technologies, allowing them more "efficient" attacks, so in consequence also Public Safety and

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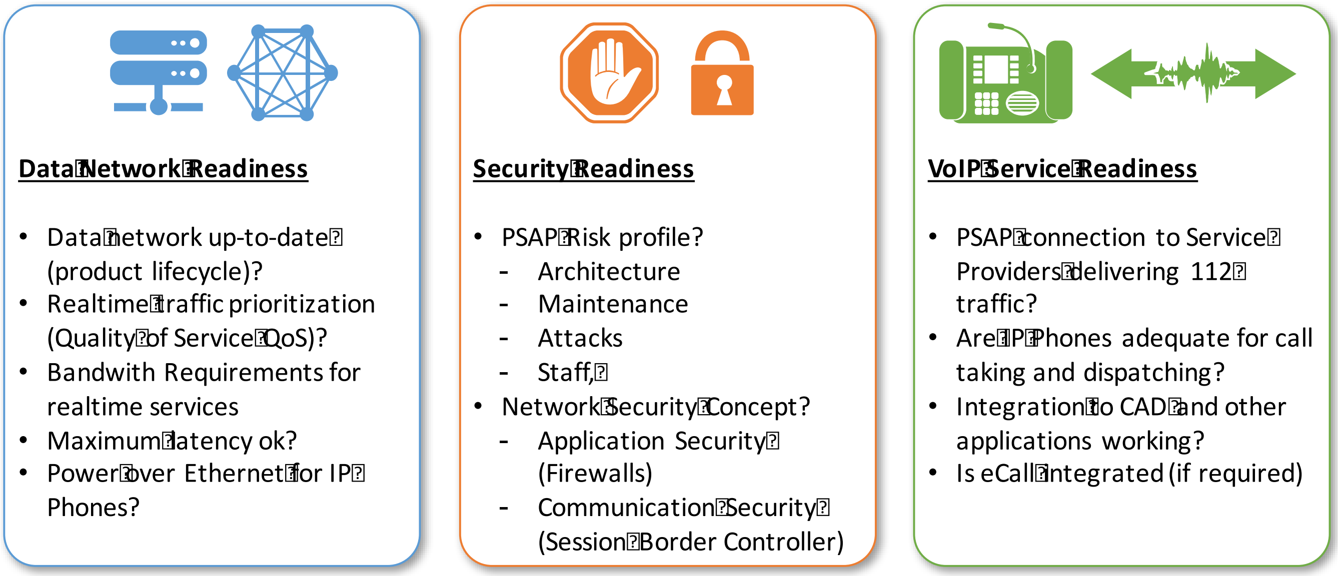
[11] As being discussed in the EENA webinar "Telephonic Denial-of-Service Attacks on PSAPs", January 31st 2017: Contingency plans, guaranteed access to emergency services even in force majeure situations through telephony networks, PSAP dimensioning taking into account large incident scenarios, education!

Emergency Services should consider to fully transition to these modern technologies in order to be able to protect themselves better against Telephony Denial of Service attacks.

4 Introducing VoIP in the PSAP

Introducing VoIP as a technology in the Public Safety answering point can be a major shift for many organisations. On the other hand it could be seen as just another step and item on the road to a full technology refresh, depending on where organisations currently are in terms of technology, and what kind of technology renewal initiatives had been executed in the past.

Under this document's focus of renewing the voice service, the following three pillars aim to summarize topics that would need to be considered with this move:



4.1 Data Network Readiness

A mandatory task in a migration and technology refresh process is to check data network readiness in terms of o multiple aspects to be considered:

- Is the PSAP data network up-to-date regarding the product life cycle of the active components?
- Is the data network set up to support realtime traffic prioritization over standard data traffic, is Quality of Service (QoS) implemented?
- Is the available bandwidth sufficient to support realtime services?
- Is network latency optimized in order to minimize voice traffic delay?
- Do the access switches support Power over Ethernet for IP Phones?

These factors should be analysed and made transparent in order to detect gaps in requirements to be filled with a network technology refresh. Appendix A.7 is providing more detailed information on these aspects.

4.2 Security and Operational Safety Readiness

Security, operational safety and stability are key requirements to achieve an uptime close to 100% for the PSAP. An in-depth assessment is highly recommended when moving from a well know technology to a new technology that needs to be introduced into the PSAP IT service production:

- Is the entire PSAP risk profile assessed and determined?
  - Architecture and fault-tolerance, resiliency
  - Maintenance contracts and service levels
  - Risk to become target of attacks
  - Staff, contingency planning and cooperation with other PSAPs
- Is a Network Security Concept in place?
  - Focus on Application Security (Firewalls)

    - Focus on Communication Security (Session Border Controller)

Whilst this document has a focus on (communication) technology migration, there always is a link to data security, as well as to operational security and safety that needs to be well determined and documented.

4.3 VoIP Service Readiness

Reflecting a move to a new communication technology, there are several aspects that have to be considered along the emergency communication chain:

- How is the PSAP connected to the Service Providers delivering 112 traffic?
    - Are multiple traffic feeds available into the PSAP to avoid the risk of a single point of failure on trunk access?
    - Are multiple providers connected to avoid the risk of a service provider outage?
- Are the IP Phones adequate for call taking and dispatching?
    - What is the level of functional requirements in terms of communication endpoints for call takers and dispatchers?
    - Will standard phones be sufficient for communication in terms of workflow, buttons to be pressed, access to specific features or services (e.g. radio integration)? Is there a need for more specialized phones for specific purposes (e.g. tradeboards or turrets), also looking at the call taker's and dispatcher's current workplace?
    - Is there a requirement to move to softphones over time in order to achieve a higher level of functionality and integration, especially with regard to the desktop applications and workflows, also see chapter 4.4 "Side Aspect of VoIP: PSAP Application Integration"?
- Is the integration to the CAD and other applications working?
    - Current integration capabilities to be maintained?
    - New requirements for renewed services?
- Is the radio integration working as expected?
    - Functional integration to directly access required radio channels or talking groups?
    - Traffic integration between the PSAP VoIP service and the radio traffic, does a migration to VoIP require upgrades on existing radio gateways or even new radio gateways to provide SIP connectivy, Radio over IP (RoIP)?
- Is eCall integrated properly (if needed for EU eCall compliance)?
    - Are any existing eCall modem deployments affected by the migration to VoIP?
    - Is there a requirement for new eCall modems?
    - TPS eCall (Third Party Services) integration, breakdown assistance access to emergency call taking

4.4 Side Aspect of VoIP: PSAP Application Integration

The approach taken in this specific document is not taking into account any dependencies between the communication layer and the application layer with its business applications

- Call taking,
- Computer Aided Dispatching (CAD),
- Geographic Information System (GIS),
- Crisis Management,

just to name the most obvious ones.

In current solution deployments the typical integration method is CTI (Computer Telephony Integration), which basically links the communication layer (PBX) to the Application Layer by handing over the calling party number of the emergency call from the phone system to the application platform, in order to be further processed there. A typical element to link communication and application is a CTI softphone, a control element for the call takers or dispatch to manage their phone function (pick, forward, terminate and make call).

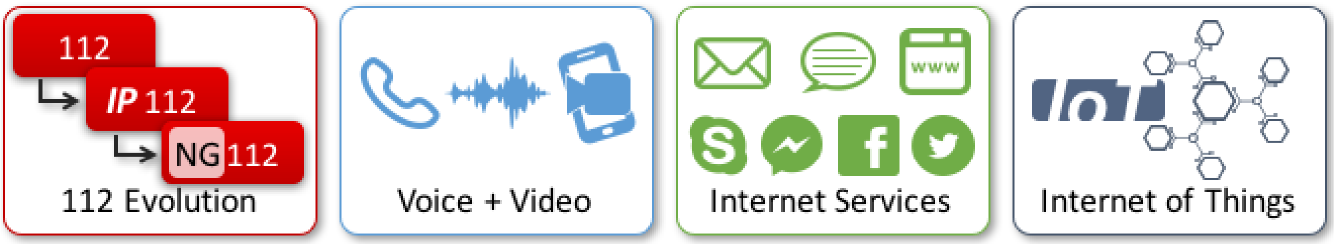
With a view to a like-for-like migration scenario from ISDN to VoIP we can expect that basic CTI connections will remain available. However, we have to consider that in some case we might find very old CAD and call taking applications that would not be ready to interwork with newer communication platforms without an update to the latest release of the application framework. This should be checked when planning a migration.

Vice versa, replacing an old ISDN PBX with a modern software-oriented communication platform, there should be new integration capabilities like web services available in order to connect the communication platform to a browser-based CAD platform. Using web services is going to offer new capabilities for tighter integrations, and with regard to "more context with SIP calls" this can open the path for enhancements on the procedural side in the emergency response organisation.

5 Strategic Outlook: What's next after transforming the voice service?

Starting the journey to renewing PSAP technologies, especially realtime communication services by introducing VoIP is a first step that should be seen under an overall strategic perspective, with the potential of being very influential beyond the area of infrastructure renewal:

- Supporting the 112 evolution towards NG112
- Adding voice as the natural next step and enabling Multi- and OmniChannel communications
- Leveraging the Internet and offering the inclusion of Internet services
- Driving automation and Digital Transformation adopting the 'Internet of Things' (IoT)



5.1 The path to Next Generation 112 (NG112) – PSAP perspective

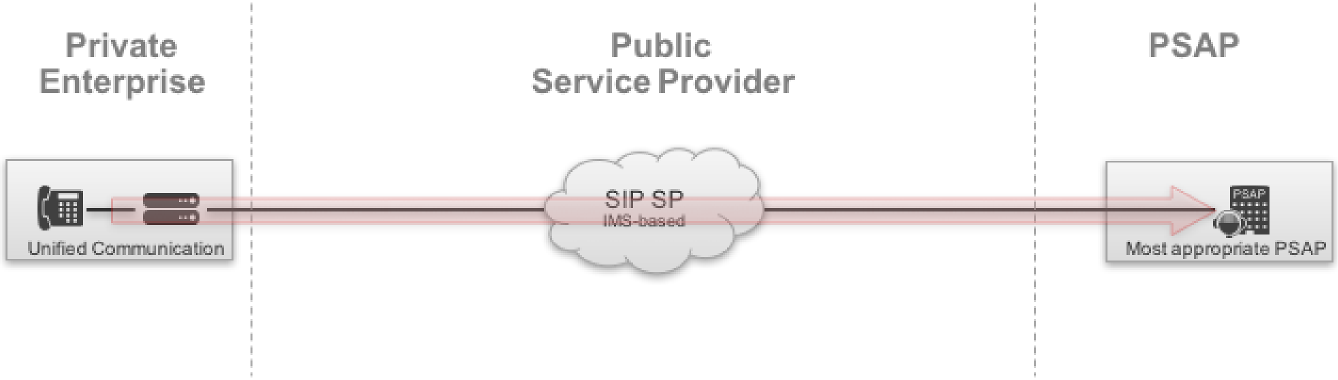
Getting ready for VoIP internally in the PSAP immediately enables the move from traditional emergency trunk access using ISDN to SIP trunks.

NG112[12] as the far-end goal in 112 evolution builds on VoIP with applying the Session Initiation Protocol (SIP) end-to-end in the emergency call chain, assuming SIP trunk connectivity for the PSAP. As standardization of

[12] in Europe also known and discussed as "Emergency Calling 2.0", "Next Generation Emergency Calling", as well as "NG 9-1-1" in the US and elsewhere

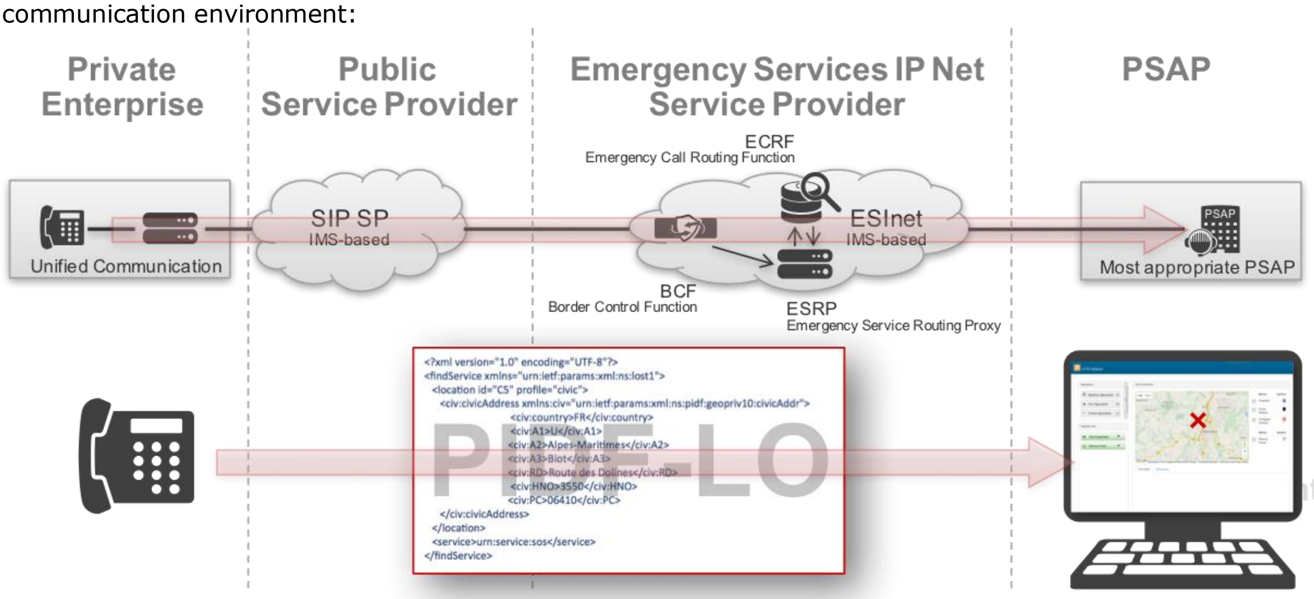
NG112 in ETSI[13] as well EU regulations[14] are still dynamically moving and changing, we cannot assume full adoption of NG112 principles[15] at the time of the technical move from ISDN to SIP with the majority of service providers in the EU, there is an immediate need to be covered to allow connection of PSAPs to public networks using the SIP protocol.

This interim step very often is referred to as “IP 112”, “IP emergency calling”, “112 over IP”, as shown in the following graphic with an enterprise emergency calling example:



So whenever moving to VoIP internally in the PSAP, “IP 112” is assumed to be available with that, or probably even being the driver for the transition.

The next step from “IP 112” to “NG 112” basically adds the requirement of reading the GeoLocation-header (and the contained PIDF-LO[16] location information) from the incoming emergency call’s SIP Invite message, resulting in the immediate availability of the caller’s location for further use in applications as well as operational procedures, as being shown in the below graphic for a call originated in an enterprise communication environment:



Describing the multi-step journey towards NG112, especially with the step from IP 112 to NG112, there should be a specific interest understand PBX vendor’s view on this journey as part of the PSAP’s decision making process on technology refresh, preventing costly replacements of IP PBX architecture for the purpose of adding the functionality required with NG112.

[13] ETSI Emergency Communications (EMTEL), Special Committee focused on standardisation in emergency communication, established in 2005

[14] Completion of standards in response to EU mandate M/493 to produce the relevant standards to support the Location Enhanced Emergency Call Service as described in ETSI ES 203 178

[15] mainly understood as transport of caller location information in the SIP header of an emergency call, as well as location- and policy-based emergency call routing within Service Provider’s networks (ESInet, Emergency Services IP Network)

[16] PIDF-LO, Presence Information Data Format-Location Object according to IETF RFC4119

5.2 Moving beyond voice: MultiChannel, OmniChannel, Total Conversation

Being prepared for VoIP as a real time application in the PSAP, other real time applications like Video can be introduced easily, as most of the work for data network readiness has already been done in terms of Quality of Service and traffic priorisation.

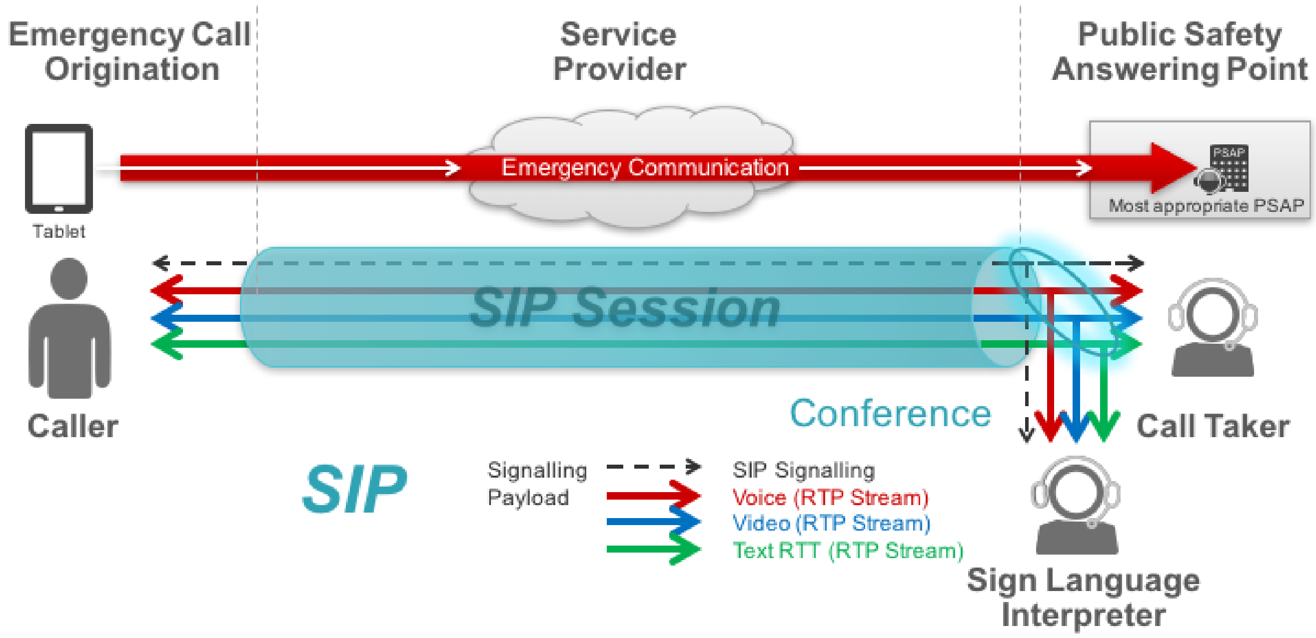
Unlike in ISDN, a video call in an IP-based environment is very similar to a voice call: it's a SIP session with media 'video' instead of 'voice'. Therefore, modern IP PBXs are very capable of handling video calls in the same way as they do with voice calls.

Of course a video call does require a different device at the call taker's desk, providing visual communication capabilities to the call taker. This in fact could be a hardware device optimized for video, or a software application on the call taker's PC screen. With the capabilities of modern headsets connecting to multiple communication endpoints at the same time, even the call taker's desk can be very clean with only a minimum number of devices and accessories.

With that kind of approached the PSAP would have turned into a MultiChannel Communication Center.

Thinking into the direction of visual oriented communication for the call taker, the topic of 'Total Conversation' immediately comes to mind. 'Total Conversation' is an approach and a concept to grant accessibility to 112 for people with disabilities[17] by combining voice, video and Real Time Text into a single communication session, based on SIP.

So by adding 'Real Time Text' (RTT) capabilities to voice and video, a kind of OmniChannel (all channels in parallel) would become realistic.



Callers with difficulties in hearing and speaking would be enabled to access emergency services by using Total Conversation apps on their smartphone or tablet, and on the receiving side in a sign language interpreter would be added to the communications, supporting the call taker or dispatcher to work on the case.

5.3 Consuming and delivering Internet-based services

Preparing the PSAP to become VoIP-enabled anyhow is been seen as a relevant next step in technology refresh initiatives in Public Safety. That being sad, securing the PSAP for VoIP services is an important additional step, also allowing for introducing of more Internet-based services like WebRTC, SMS/Text, Messaging or Social Media, opening up for capabilities to add additional content and context to emergency communication.

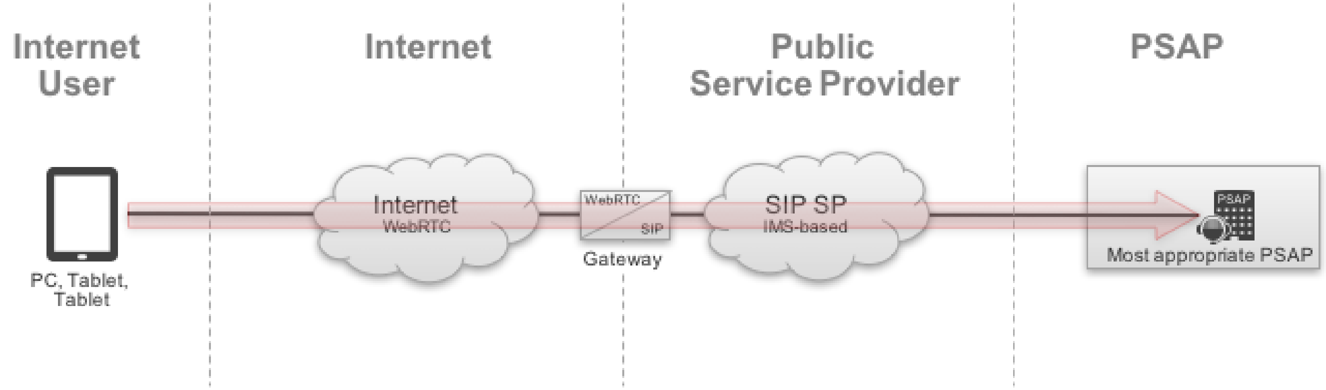
[17] EENA Operations Document "112 Accessibility for People with Disabilities", released 2012
http://www.eena.org/uploads/gallery/files/operations_documents/2012_01_13_112accessibilityforpeoplewithdisabilities.pdf

With the risk level of even more threatening experiences like terrorist attacks in our society, the unavailability of telephony services at all due to blocked mobile phone networks in the affected area, congested transmission networks on the way to the most appropriate PSAP, and overwhelmed PSAP staff at the end of the chain, organization in Public Safety and Emergency Services need to think about how cope with the changing landscape. 112 as a well known 'brand' for emergency response is driven beyond the limit of its capabilities and effectiveness under certain conditions[18], so new ways of emergency communication need to be discovered, and probably extend 112 from the voice domain into the Internet, making Digital Transformation a reality here as well. This means embracing all obvious channels, including the most popular internet-based real time services like e.g. Skype and WhatsApp as well as embracing social media services like Facebook or Twitter.

5.3.1 Alternative realtime services to voice calls

Some of these services are technically very closely related to Voice over IP, and gateways transforming a communication through WebRTC[19] originated on a web page, Skype or WhatsApp can extend the reach of the 112 to other popular channels that are currently not connected to 112[20], but are received as a kind of phone service by the citizens.

The Internet itself does offer more matured channels for realtime communications from within web applications. WebRTC as a way to enable browser-based voice and video has arrived in the industry, and the concept of a national web page like www.112.de can deliver interesting approaches of giving 112 voice and video access to people with non-phone devices (e.g. tablets on WiFi networks), which could also make use of the location information available through the device's Geolocation API[21] to precisely locate the caller. That being said, leveraging a gateway function between the WebRTC-oriniated voice traffic from a web page and the PSAP could bring this traffic into existing VoIP and Video over IP deployments to be answered, as shown in the graphic below:



[18] EENA Conference 2016 session: "Use of social media during Paris attacks" on YouTube https://youtu.be/V3NudseyiZE?list=PLAuBrNEvppxFMXCKTcTim4zus1Z0_EQx3
[19] WebRTC: Web Realtime Communication, voice and video communication from within a web browser
[20] and are legally not required to connect to 112 as they are not considered a public service provider for voice services
[21] Geolocation API: Application Programming Interface on smart devices to determine GPS location

5.3.2 Non-realtime services in relation to realtime voice services

As this document is focused on Voice over IP in relation to 112, it still seems valuable to mention that other services like Social Media that are currently widely discussed can become available as a side effect of renewing the existing voice communication platforms. Many vendors do offer access to non-voice and non-realtime services as an integral part of their unified communications and customer services platforms, so this should be considered when planning for the future.

Reviewing what has happened during the most recent terrorist attacks in Europe[22], we can observe the same pattern everywhere: voice services being overwhelmed, people in distress and danger moving to Facebook and Twitter to become vocal and communicate again, and also the Police picking up outbound communication on Twitter as an effective means to get guidelines and immediate ask for citizen's actions published.

But still as we see this kind of emergency communication on Social Media in most of the cases staying silo'd communication, decoupled from the voice communication world, far from being integrated and highly automated. It could be very beneficial for public safety and emergency services learning and picking up experience from customer services organisations that have been mastering Multi- and OmniChannel already. The required solutions are commercially available, and many of the traditional IP PBX vendors are in the position to extend communication capabilities to other channels as well.

5.4 Automation and the Internet of Things (IoT)

Currently there are a lot of conversations around the benefits of the 'Internet of Things' (IoT) for Public Safety and Emergency Services[23].

The overall promise of IoT is in two areas:

- Leveraging intelligent devices, equipped with sensors and attached to the Internet in order to raise alarms automatically and to provide additional context, e.g. temperature, humidity, smoke/gas. These are typically scenarios discussed in conversations around smart cities and smart homes.
- Using IoT-approaches for the purposes of managing emergency situation[24], e.g. to embed drones into emergency response procedures and use them as a communication endpoint es well.

Introducing VoIP based on SIP opens the path to more automation and IoT-based services, as many mechanism are expected to be used in IoT-related communications as well, please refer to the EENA Technical Committee Document "Public Safety Digital Transformation – The Internet of Things (IoT) and Emergency Services" for more details.

5.5 eCall as an enabler and driver for change

Finally, to shed a light on to eCall at this point and put it into context of the current development, we can consider eCall as an enabler and driver for many other new services in public safety, because it's the first new 'channel' that appeared and was demanded by EU legislation to be adopted. Whilst still being a voice call, initially designed to be delivered through classic telephony networks, eCall can be looked at even more from an IoT service's point of view rather than a simple voice call.

- eCall is an automated call
- eCall is triggered by sensors
- eCall connects a "thing" (car) to public safety
- eCall comes with additional data providing a deeper situational context

eCall in fact is urging emergency response organisations to adopt their well established procedures to the new characteristics of a non-voluntarily non-human originated call and situational insights delivered with the information embedded in the eCall MSD[25].

[22] Paris Bataclan shotting 15th November 2015, Berlin Christmas Market 19th December 2016
[23] See EENA Technical Committee Document "Public Safety Digital Transformation – The Internet of Things (IoT) and Emergency Services", published 2016 http://www.eena.org/download.asp?item_id=170
[24] See EENA Operational Committee Document "Remote Piloted Airborne Systems (RPAS) and the Emergency Services", published 2015 http://www.eena.org/pages/drones#.WJx1xhiX-qA

We can consider eCall as an IoT-Service in disguise of a voice call, and once emergency response organisations have fully mastered the handling and management of eCall, they will be more prepared and open to other innovations as well.

6 EENA Recommendations

Stakeholders	Actions
European Authorities	1. Drive Digital Transformation in Emergency Services and Public Safety, re-thinking and renewing current 112 services in accordance with demographic change, citizen expectations and technology development. 2. Foster cross-border collaboration between EU member states. 3. Define data privacy requirements to support additional data associated with NG112 emergency communication
National Government	Review current emergency services legislation in terms of federal vs. national responsibilities in order to enable and drive better collaboration between authorities, across regions and areas, as well as cross-border between EU member states and other countries.
National / Regional Authorities	1. Define methods and requirements for inter-agency data exchange and real time communications 2. Start cross-agency collaboration pilot projects with focus on seemless data exchange and realtime communications.
Emergency services	1. Start strategic planning to include Internet-based services into communications and security architectures. 2. Focus on VoIP access to network operators as the first step to be implemented.
National telecommunication regulator / Network Operators	1. Drive technology migration from traditional telephony to VoIP and SIP-based access services, define the standards to be applied between network operators and PSAPs. 2. Assess NG112 concepts and standards, propose and develop country-specific adoptions based on national requirements and capabilities.
Network Operators	1. Implement national regulator's guidelines for emergency services access to the public network with VoIP. 2. Assess NG112 concepts and standards to adopt and embed required functional elements into their IMS networks.
Standardisation bodies	1. Develop "Next Generation 112" to become an accepted standard. 2. Specify and develop more use cases beyond simple voice calling to 112, extend to other non-voice internet based channels (video, text messaging, social media, automation/Internet of Things)

[25] MSD: Minimum Set of Data, a structure of 140 Byte containing information about e.g. the car's location Vehicle Identification Number VIN, propulsion type

Appendix A Technology Explained

This chapter outlines the basic technologies that are used to build telephone networks, with the objective to give interested non-technical readers an insight in which way these technologies are enabling voice communications and how they have been developing over time.

In principle the concepts of 'Circuit Switching', representing the traditional technologies in telephony, and 'Packet Switching', being the basis for 'Voice over IP' will be explained in order to allow building an understand of how transport of a voice connection over a data network works and what the challenges embedded in this approach are.

The following sections after that describe a more infrastructure oriented view 'Voice over IP', with a look at Local Area Networks and the capabilities they need to provide as well as the necessary network engineering guidelines to enable high quality voice services to be delivered.

A.1 Circuit Switching, part 1: Analog Telephony

Telephony or voice services from when they were made accessible to the public until the late 1990s have been delivered on analog or digital communication systems, working according to the principle of 'Circuit Switching'. This term describes the way that connections are established in order to allow communication between people:

- phones are connected to the telephony system by a pair of copper wires. Whenever those wires are connected to some kind of loop with the telephones and other devices being part of that loop, this is called a 'circuit'.
- telephony systems, called an 'exchange', themselves use copper wires internally for every single line that they can connect to as well. They have complex cascaded mechanic switching fabrics internally to connect (or 'switch') between these different lines
- large telephone systems deployed within public service providers networks are called 'Exchange Offices', private enterprise telephone systems are called 'Private Branch Exchange' (PBX)
- connections between telephone systems are established as well using copper wires, bundled into cables of bigger capacity
- in order to establish a connection from one phone to another these different segments of wires are connected through the switches internal to the various telephony systems, controlled by the internal logic that acts according to the dialed number, resulting in a 'switched circuit' which then is a successful electrical connection between these two phones.

The following graphic illustrates the principle of circuit switching:

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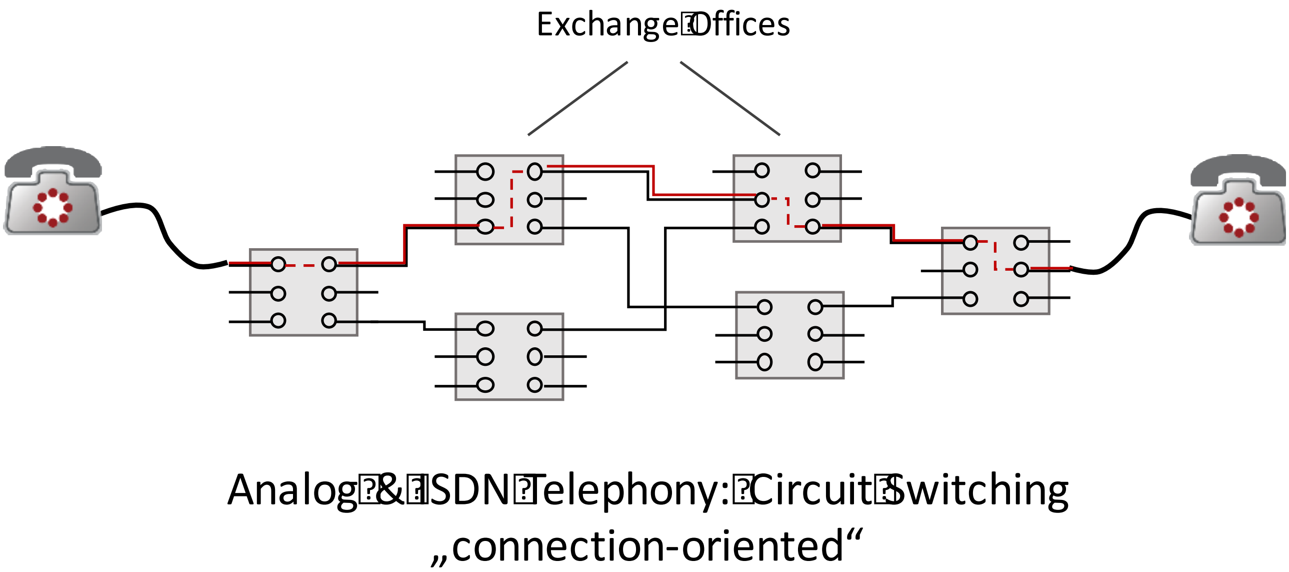

In the analog telephony world, a physical telephone line is requesting a pair of wires as transport medium, and a phone call occupies one telephone line completely for the duration of the call. So the number of possible simultaneous phone calls between two points equals the number of wire pairs.

A.2 Circuit Switching, part 2: ISDN telephony and Time Division Multiplexing (TDM)

As a next step in technological evolution, the wire- and electricity-driven principles of phone services have been brought to a higher level of efficiency, allowing more concurrent connections, introducing the first wave of digital communications. The principle of 'Time Division Multiplexing' (TDM) moves the phone call from the analog domain into the digital domain and allows to multiply the usage of a single telephone line for more than just one connection at a time. The following graphic illustrates the TDM principle, which was the basis for the introduction of ISDN in the early 1980s. In TDM, the transported "voice content" is cut into small "chunks" and packaged into small identically-sized segments of data. Looking at the graphic below, different phone connections are represented by different colours (green, red, blue, yellow), and they are carried across a single connection, one after the other. On the receiving side, the original sequence of segments is re-assembled again, and again a constant flow of the "voice content" is generated. In a phone call, these segments are sent constantly, independent from what "content" they carry, a part of a spoken word or just silence while no one speaks, so the efficiency of the offered transport capacity shows potential for optimisation in terms of maximum capacity.

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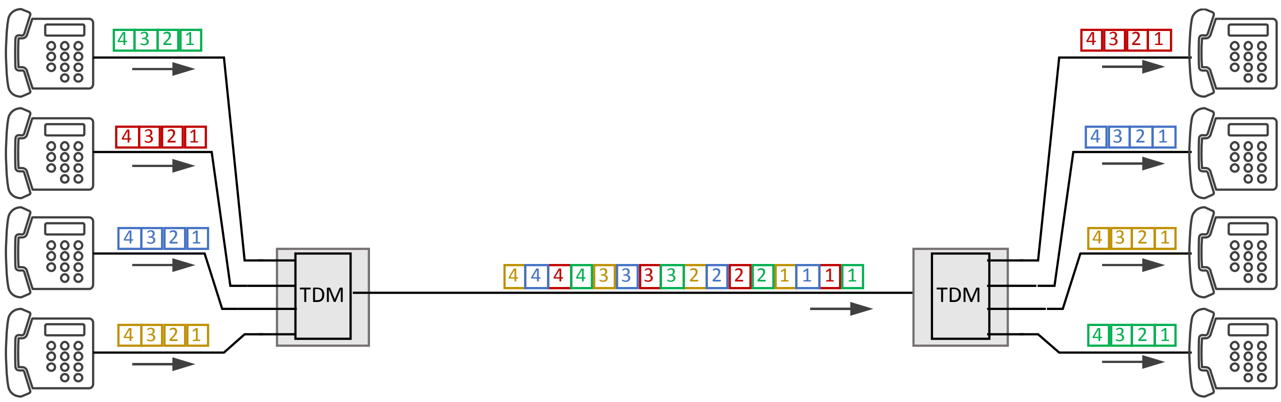


Digital „ISDN“ Telephony: Time Division Multiplexing (TDM), Circuit Switching „connection-oriented“

Both technologies, analog as well as ISDN digital circuit switching are closed technologies, designed and built for the purpose of voice communication only. Beyond that, the only other services that have originally been implemented on this kind of technology were those that also make use of audible tones in order to transport information, like fax and modem services.

This also means that that the requirement for resources and capacity is also constant for every service, no difference between multiple calls or within a call. These are the characteristics of what we call a synchronous service, both sides of the connection as well as all switching elements in between are completely in sync, guaranteeing a very low delay of the transported information, which is needed in order to achieve an acceptable speech transmission quality. A human's hearing and processing apparatus (ear and brain) are very sensitive to the constant flow of information, especially long delays on a call make communication between two people very hard or even impossible.

With a digital phone connection over ISDN, the voice signal is passed from the calling party to the called party inside a synchronous data channel of 64 kbit/s. This bandwidth is declared in the ITU[26] ISDN standards (using G.711 linear voice encoding), and is assumed to deliver a high voice quality. In fact, ISDN quality has become the de-facto quality measure for telephone communication over time.

A.3 Packet Switching: Internet and Voice over IP (VoIP)

In the late 1980s, in parallel to the development of digital voice communication, data networks became more and more popular. With data networks, a different way of transporting information occurred, called 'packet switching'.

A 'packet' consists of header plus data. Each packet has signaling information (source and destination addresses) in the header. Compared to telephone principles this means that every single packet "dials" the number of its destination. This signaling information is used by the intermediate exchange devices to route each packet individually ('connection-less', compared to 'connection-oriented' in an analog or ISDN phone network).

[26] ITU: International Telecommunication Union, international standards body

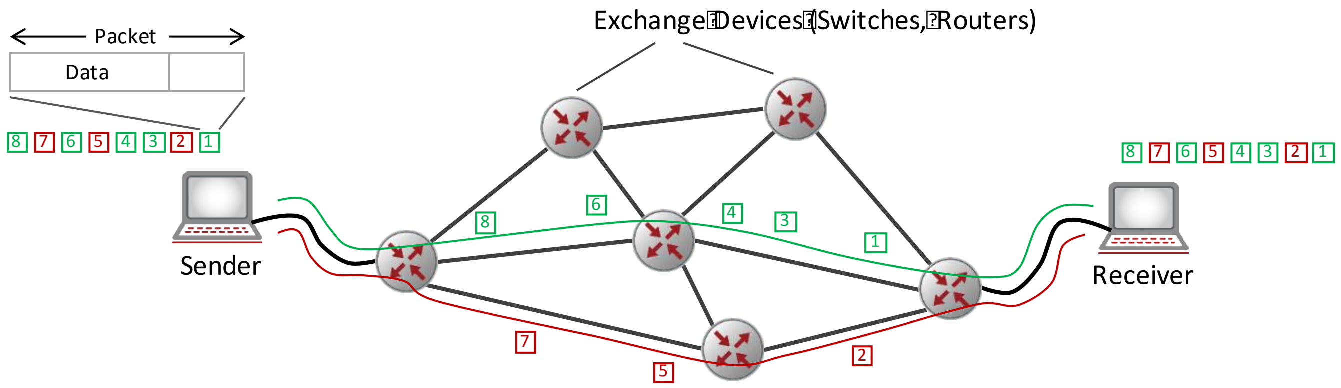
Different packets can use different routes from sender to receiver, the delay on the transmission of a whole sequence of packets could be non-constant, this is the characteristic of an 'asynchronous service'.

For pure data transmission synchronicity in transmission is not necessarily needed, because other than in a voice service for an email or a web browsing service the person that is using this service does not experience this delay or un-constant transmission (called 'jitter'). The only thing he could experience is that the email might arrive 2 seconds later or the web page takes a little longer to build.

The utilization of transport capacity with packet switching is quite efficient, a packet is only sent when it really has something to transport. Other than in phone calls, there is no end to end reservation of transport capacity, so multiple connections simultaneously share the same link, whenever they want, without upfront warning, so there is a kind of competition between different services or applications with regard to the transport capacity.

The graphic below shows the principle of packet switching:

- packets created at the sender's side to carry data to the receiver are numbered according to the sequence of packet creation
- they are transferred over a network of exchange devices, called routers and switches, where these exchange devices potentially chose different paths across the network for different packets (in the example the green and the red routes)
- As each of the routes can have different characteristics on terms of speed and latency, the packets most probably will not arrive at the receiver's exactly in the order they have been sent, so the receiver will need to be prepared rebuild the original sequence based on the packet numbering



Data Networking: Packet Switching
„connection-less“

Data networks applying the principles described above have been built by different vendors, using different implementations and protocols. These implementations were Local Area Networks (LAN) to facilitate data exchange between devices on a campus.

In the late 1960s universities in the US started to connect different Local Area Networks, developing a new technology and a new protocol for that purpose, the "Inter-Network Protocol", which nowadays is the basis for the global service called the 'World Wide Web' (WWW) or simply the 'Internet'.

The Internet in fact is not a single network, it is a network of networks, coupled by a number of important protocols , in which the most famous one is the 'Internet Protocol' (IP), currently mostly used in its version 4 (IPv4). In order to identify a device on the Internet, a special form of addressing schema is used, known as the Internet Protocol Address (or IP Address), in its typical shape of four blocks with three digits each (e. g. 192.168.001.243), which is the equivalent to the telephone number in a telephone network.

Currently there a strong push to move from IPv4 to IPv6, mainly to overcome the restrictions of limited number of IPv4 addresses. In the days of the development of IPv4 the numbers of computers or network devices that could potentially connect to the Internet has been seen in a different light, compared with today's perspective with the 'Internet of Things', appearing on scene to basically allow connecting every device to the internet: PCs, tablets, smartphones, TV sets, wearables, medical devices, sensors, refrigerators, coffee machines, and probably many more in the future, going beyond our current imagination.

In order to match voice quality expectations (see ISDN voice quality with 64 kbit/s in the previous chapter), also a phone connection over IP packed switched network can use the G.711 voice encoding. As a matter of fact, in IP networks a net bandwidth of 64 kbit/s to be delivered end-to-end does require approximately 100 kbit/s of consumed bandwidth according to protocol overhead.

Beside G.711 there are other codecs with the same, higher or lower bandwidth requirements to accommodate the bandwidth available in different environments, also resulting in different perceived voice quality. The most frequently used standards in professional VoIP systems are:

- G.711, 64 kbit/s, high quality voice
- G.722, 64 kbit/s[27], 'Wideband Audio', higher voice quality[28] than G.711
- G.723, 5,3 kbit/s[29], 'Lowband Codec', lower voice quality than G.711
- G.726, 32 kbit/s, similar voice quality to G.711
- G.729, 8 kbit/s, lower voice quality than G.711

Not all codecs are supported on all VoIP platforms.

There are also other lowband codecs that are mainly used in Internet-based voice services like e.g. Skype. These codecs have been developed later than the above mentioned codecs, typically resulting in a higher bandwidth efficiency compared to the achievable voice quality.

At this point it should be mentioned that there is a difference between "Voice over IP (VoIP)" and "Telephony over IP (ToIP)":

- **Voice over IP (VoIP)** is the more commonly used term. It's reflecting the mechanism of transporting a voice service over a packet-switched network, not specifying any further to what kind of 'packed switched network' referring to.
- **Telephony over IP (ToIP)** is referring to "business" or "professional" use of VoIP in a controlled and controllable environment in terms of "Quality of Service" (also see chapter A.7 "Voice Readiness of IP networks – Best Practices").

That being said, an Internet-based service like "Microsoft Skype" for consumers[30] with no end-to-end "Quality of Service" would be called "VoIP", an enterprise deployment of professional PBX systems, attached to a Service Provider's SIP trunking service would be rather referred to as "ToIP". As mentioned above, the more commonly used term in the industry is VoIP. So when referring to "VoIP" in this document, the assumption is professional and business-grade usage, unless mentioned otherwise.

[27] 56 kbit/s, 48 kbit/s alternatively
[28] at 64 kbit/s
[29] 6,3 kbit/s alternatively
[30] not to be mixed up with „Microsoft Skype for Business"

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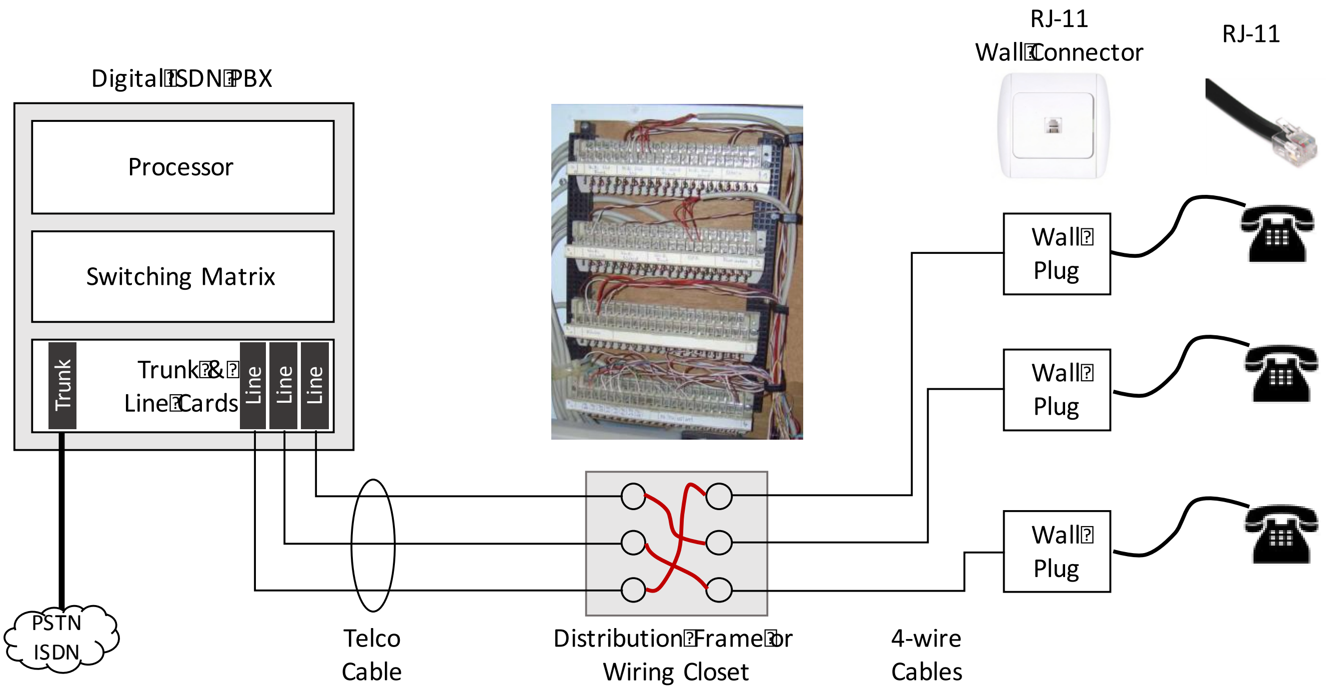
A.4 Cabling for Telephones

Desktop Telephones as the most relevant endpoint in voice communication in the PSAP need to be connected to the PBX via cables. Depending on the technology used, analog/digital/ISDN telephones versus IP telephones, different connectors and cables will be used, also requiring different "cable backbones". Consequently, these "backbones" get mainly into focus when moving from ISDN to VoIP on the telephone side.

A.4.1 Analog/digital/ISDN telephones

Analog, digital and ISDN phones are basically all connected in the same way to the PBX:

- The phone on the desk itself is plugged into a wall-mounted telephone jack, using a flexible 4-wire copper cable with an RJ-11 connector on both ends.
- Wall connectors are attached to individual 4-wire cables – crimped, screwed or soldered – that run from the connector to a distribution frame or wiring closet.
- In smaller environments there typically is only one wiring closet involved in the path between phone and PBX (typically very close to the location of the PBX). In larger environments (multiple floors, multiple buildings), we usually see more than one wiring closet or distribution frame, e.g. one per floor, to collect the individual cables from the phone and to consolidate them into higher-pair cables, reducing the effort to pull a high number of single cables through the building.
- Between the PBX and the distribution frame usually there is a telco cable that connects the wires to the line cards in the PBX.
- A single line card usually can connect 8, 16 or 24 phones to the PBX.



A.4.2 IP telephones

IP phones are all connected in the same way to the VoIP PBX, using the regular data network with Ethernet cabling everywhere:

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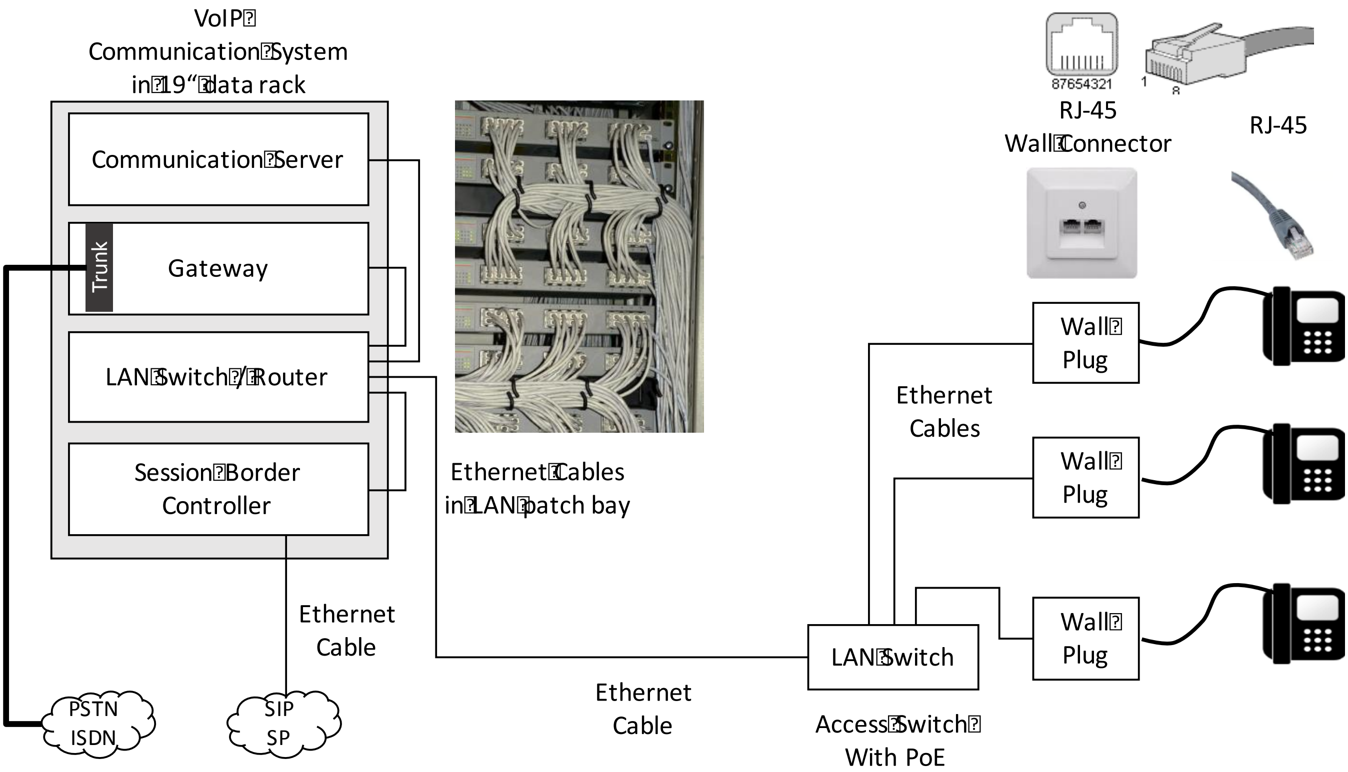
- The phone on the desk itself is plugged into a wall-mounted Ethernet jack, using a flexible 8-wire Ethernet cable with an RJ-45 connector on both ends.
- Wall connectors are attached to individual Ethernet cables – typically plugged – that run from the connector to a LAN switch. This switch is called an "access Switch", as it is giving access to phone. As IP phones need to powered by the LAN, this switch needs to be equipped with Power over Ethernet capabilities. Typically access switches come with 12, 24 or 48 ports.
- The access switches from the different floors or buildings are connected to the central data room's equipment Ethernet cabling, typically to a core switch that also contains an IP routing function in order to connect traffic to different subnets and towards the outside world.
- Very large environments typically have another network layer of distribution switches between the access layer and the core layer.
- In very small environments, access and backbone switches could potentially collapse into a single unit, which of course then has to provide PoE to the IP phones as well.



According to the growth in Ethernet transmission rates (see next chapter), also cable specifications have changed over time to adopt to higher frequencies and thus higher transmission rates and greater cable lengths.

Copper-based Ethernet cables are classified in categories, basically reflecting the data transmission speed they can handle. The most commonly used cable categories to support VoIP are

- Category 5: 100-Mbit-Ethernet
- Category 6: 1-GBit-Ethernet
- Category 7: 10-GBit-Ethernet

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There are also optical fibre Ethernet cables that are used especially to connect active network elements like switches and routers over longer distances than copper cables are able to do.

A.5 LAN technologies: Ethernet and Wireless LAN

The Internet Protocol itself, managing addressing and routing between elements and components in the network, needs a transport technology in the Local Area Network to run on. The dominant technology is the 'Ethernet', well known by it's Ethernet cable. Ethernet cables are plugged into 'port distributors' like Ethernet Hubs and Ethernet Switches using an RJ-45 plug to give endpoints like PCs and IP telephones access to the network with a number of ports.

The international standards associated with Local Area Networks are collected under the IEEE 802.X-series of standards.

A.5.1 Ethernet

A typical Ethernet cable consists of four copper wires, each two for the transmitting (TX) and two for the receiving (RX) direction. Originally, the first Ethernet ever was a shared medium – just like radio waves in the 'ether' – where all devices where mechanically cramped to the same cable (yellow cable, according to its initial colour), sharing this cable as their common transmission medium. Every device on that cable had to listen into the 'ether' before sending a packet in order to prevent collisions of data packets being sent at the same time. The maximum transmission rate in these days was at 10 Mbit/s.

An Ethernet Hub is a relatively dumb network element, basically introduced to be able to connect devices more easily than attaching them to a yellow cable. Simply said, a hub is plugging all of the copper wires from the cables together. So all devices connected to a hub still share the same pairs of copper wires to transmit or receive data, competing for the transmission media like in the days of the yellow cable. Ethernet hubs are quite cheap network elements, but on the flip side, too many devices connected to a hub, or devices with a high amount of traffic, will result into traffic collisions, and thus limit the amount of transmission capacity.

An Ethernet Switch is more intelligent than a hub, and thus also more expensive. An Ethernet Switch allows to move away from 'shared media' like in the early Ethernet days to 'switched media' by introducing an intelligent backplane in the switch, basically by having more hardware transmission channels available at the same time. Switching technology started a race for speed on Ethernet connections, moving from 10 Mbits/s to 100 Mbit/s, to currently 1 Gbit/s and 10 Gbit/s seen as the next step on a single cable. Professional Ethernet switches typically come with 24 or 48 ports to connect to the different end user devices.

This increase in bandwith and transmission capacity enabled real time communications like voice and video to become available on data networks.

A.5.2 Power over Ethernet (PoE)

In the days of analog or ISDN telephony, no one was really thinking about how to power a telephone. Power was just there, once the phone was connected with its two-wire cable. So in fact powering the phone itself was a capability of the enterprise telephone system (PBX) or of the local exchange office, and the power injection into the phone work because of the copper cables connected to batteries or power supplies through the wire's 'loop'.

Now with VoIP the phones are no longer directly attached to any phone system, but to an Ethernet switch, thus there is no such concept of 'loop' any more. In order to avoid having to equip every IP phone with an extra power supply, Ethernet switches started to include this capability.

Power over Ethernet (PoE) is a required service in Ethernet switches that shall be used in the access layer of the network, because many of the devices that connect to them, such as IP phones, but also wireless access points, video surveillance cameras, and digital signage, will require the service in order to operate.

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The ability to centrally power these devices from the access layer not only makes the physical installation simpler, but it also provides a central point to provide power resiliency to all of the devices through a Uninterruptable Power Supply (UPS).
The set of standards covering PoE are collected under IEEE 802.3, where the most recent standard is 802.3at, also known as PoE+. PoE+ supports an increased power draw over previous standards, up to 30 watts, which allows for a more diverse set of devices that are able to utilize the service, as well as for the advancement of modern WLAN technologies including 802.11ac (see below).

A.5.3 Wireless LAN

Besides cable-bound Ethernet the evolution of affordable wireless data services started in the 1990s, called Wireless LAN or simply WLAN, also known as WiFi in the space of consumer's products.

Like in the wired age, also in WLAN standards the race for speed is on. The first commercially available WiFi products used the IEEE 802.11a and b standards in the late 1990s came with a maximum transmission rate of 5 Mbit/s. Later on 802.11g, 802.11n and currently 802.11ac became available, with transmission rates reaching beyond 1 Gbit/s depending on actual products used and the configuration of the WLAN Access Points.

Also here we see that for mobile devices with WiFi capabilities realtime voice and video services over IP networks become a natural part of the applications and services they can expose to the users.

A.6 Why is VoIP different from Analog and ISDN Communication Technologies?

With an understanding of packet switching versus circuit switching we now can identify the main difference between classic voice networks and classic data networks.

A voice service requires:

- a (perceived) constant bandwidth between sender and receiver
- a low delay to avoid echoes and difficulties in hearing and talking
- a very low loss of data as well as a fixed order in the flow of packets in order to avoid drop-outs and crackling noises in the audio flow

In principle, the initial data networks were not built with these characteristics in mind. On the other hand, the development of transmission capacity was so fast that the idea of filling this capacity with voice (and also later with video) was born in the 1990s.

As a consequence, the technology of a data service needed to be prepared and trimmed in order to allow a high quality real time voice service, a synchronous service that runs on an asynchronous network!

As a result, standardization bodies began to outline requirements and develop standards allowing a data network to become 'voice-ready', as well as defining protocols that make the capabilities of voice networks, especially ISDN, available in data network-based communication platforms.

Resulting from this development, different signaling protocols to exchange commands between the central telephony system (the PBX) and the phones have been developed, which are discussed in the following subsections. Until now, all of these approaches rely on the same transport mechanism to stream the media (the voice or video signal) associated with the sessions over the IP network: the Realtime Transport Protocol (RTP)[31] or its encrypted variant Secure RTP (SRTP).

A.6.1 H.323 and vendor-specific protocols

[31] RTP on Wikipedia: https://en.wikipedia.org/wiki/Real-time_Transport_Protocol

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For the first step of telephony becoming an application on data networks, the standardized ISDN protocols as well as the vendor-specific protocols were transported into the IP-world by a set of global standards defined by International Telecommunications Union's Telecommunications groups (ITU-T). The mostly known standard here is H.323 (mainly focusing on the way phones communicate with the PBX), which then had vendor-specific additions in order to provide vendor's specific sets of features that their customers were used to also in VoIP solutions.

It turned out to be that H.323 in fact was quite a "heavy" stack of protocol in terms of how it is implemented using binary coding as well as its lack of flexibility, it was still very close to the complex implementation of the ISDN protocols.

Looking at how web applications have emerged from the early 1990s on, it was found that the mechanisms used there should be inspected and translated into real time communications as well. 'Calls' became 'Sessions', and a 'Session' can be much more than a 'Call'.

A.6.2 Session Initiation Protocol (SIP)

Session Initiation Protocol (SIP) is an endpoint-oriented messaging standard defined by the Internet Engineering Task Force (IETF). SIP is a text-based protocol, similar to other Internet-related protocols like HTTP (HyperText Transfer Protocol) and SMTP (Simple … Protocol), for initiating interactive communication sessions between users. Such sessions can include voice, video, instant messaging, interactive games, virtual reality, and probably many more.

Due to the text-based construction of the protocol SIP is quite easy to read and understand, and it offers a high level of openness, as it has the potential to easily be extended by adding new capabilities with well known mechanisms like XML.

In order to connect different SIP-based systems to each other, SIP trunking technology is used. In order to allow interoperability, SIPconnect as an initiative with a Technical Recommendation and an industry-wide, standards-based approach to direct IP peering between SIP-enabled IP PBXs and VoIP service provider networks has been introduced by the SIP Forum[32].

A.6.3 Web Realtime Communication (WebRTC)

"WebRTC is a free, open project that provides browsers and mobile applications with Real-Time Communications (RTC) capabilities via simple APIs."[33] It is a collection of communications protocols and application programming interfaces (API) that enable real-time communication over peer-to-peer connections, allowing web browsers to not only request resources from servers, but also allowing real-time media (Voice, Video and Data) to flow between browsers of other users. This in fact enables applications such as video conferencing, file transfer, chat, or desktop sharing from browser to browser without the need of either internal or external plug-ins. WebRTC gateways provide interoperability between WebRTC endpoints and other VoIP or legacy PSTN/ISDN based systems.

WebRTC is being standardized by the World Wide Web Consortium (W3C) and the Internet Engineering Task Force (IETF) [34].

Compared to other approaches Voice over IP like H.323 or SIP, WebRTC shows some similarities, but also some fundamental differences:

- For the transport of the media (the voice and video) DTLS-SRTP is used meaning that WebRTC media is always encrypted. According to the development of more bandwidth-effective encoding capabilities

[32] SIP Forum webpage: http://www.sipforum.org/sipconnect
[33] according to http://webrtc.org
[34] according to Wikipedia: https://en.wikipedia.org/wiki/WebRTC

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the use of the open source Opus[35] codec as an alternative to the previously mentioned G.711 codec was introduced, with wideband capabilities and more sophisticated features to manage best possible speech quality over the internet in combination with lowered bandwidth requirements.

- The signaling protocol used by WebRTC applications to establish calls is not defined by the WebRTC standards and is left as a decision for the application developer. This is possible because WebRTC follows a web model in which the WebRTC endpoint connects and downloads code from an application specific Web Server over HTTPS.

Interworking between a WebRTC client on the caller's side (e.g. Internet-originated emergency call) and a H.323- or SIP-based VoIP implementation in the PSAP can be achieved using a WebRTC-gateway.

At this point, we would like to point to the paper "WebRTC and Emergency Services" published by EENA in 2016 in order to get a deeper understanding of the associated use cases in Emergency Services and the technology behind[36].

A.7 Voice Readiness of IP networks – Best Practices

Over time, with about 20 years of industry experience in deploying voice services on IP-based networks, there is quite an understanding of what the criteria are that IP networks have to meet in order to be capable of carrying voice traffic. The following characterics can be seen as a kind of best practice. However, specific vendor's recommendations may vary depending on their respective products.

A.7.1 Delay

One-way delays in excess of 250 ms can cause the well-known problem of talk-over. This problem occurs when both parties talk at the same time because the delay prevents them from realizing that the other person has already started talking. In some applications, delays less than 150 ms can impact the perceived quality, particularly in the presence of echo.

A good suggestion for the network delay between endpoints, not including the IP phones would look like this:

- 80 ms delay or less for best quality
- 80 ms to 180 ms delay can give business communication quality as perceived in ISDN networks. This delay range is better than cell-phone quality if echo is properly controlled and, in fact, is well suited for a majority of businesses.
- Delays exceeding 180 ms might still be quite acceptable depending on expectations, analog trunks used, codec type, and the presence of echo control in endpoints or network equipment.

A.7.2 Jitter

Jitter is the statistical average variance of the arrival time between packets received from the IP network. To compensate for jitter, a de-jitter buffer is implemented in VoIP telephones. The purpose of the jitter buffer is to hold incoming packets for a specified period of time such that voice snippets (or 'samples') can be played out at a regular rate to the user. In doing so, the jitter buffer also adds packet delay.

Excessive jitter might also result in packet discard creating audible voice-quality problems when the variation is greater than the jitter buffer size.

A value of 20 ms or less in general is recognized for the best voice quality.

A.7.3 Packet Loss

[35] http://opus-codec.org/
[36] EENA: http://www.eena.org/download.asp?item_id=208

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Packet loss occurs when packets are sent but not received or are received too late to be processed by the telephone's jitter buffer. Too much delay or packet misorder can be perceived as lost packets. The network might appear to be losing packets when, in fact, the packets have been discarded intentionally because of late arrival at the endpoint. IP networks are characterized by unintentional packet loss in the network as well as by discarded packets in the jitter buffers of the receiving endpoints.
Packet loss can be bursty or more evenly distributed. Bursty packet loss has a greater effect on voice quality than distributed loss. Therefore, a 1% bursty loss will have more adverse effect than a 1% distributed loss.

The maximum loss of IP packets (or frames) between endpoints should be:

- 1% or less for best quality
- 3% or less for Business Communications quality (which is much better than cell phone quality)
- More than 3% is acceptable for voice but can negatively impact on signaling between the different components of a VoIP telephony system, which can degrade voice quality also

A.7.4 Full Duplex Traffic

When making a phone call, people expect to be able to talk and listen at the same time, just like in a face to face conversation. Translated into the world of technology, this is what we call 'Full Duplex Traffic', sending and receiving data at the same time. Not all components that we can find in a data network do support this behaviour (e.g. Ethernet hubs) and only work in Half Duplex mode, either sending or receiving data at a given time. So therefore it is mandatory to install Ethernet switches with full duplex capabilities in order to build a voice capable data network.

A.8 Best practices for engineering a VoIP network

Industry best practices mandate that a network should be designed with consideration to the following factors, especially with a focus on Public Safety and Emergency Services:

- Reliability and redundancy: no single point of failure, duplication of key and core building blocks
- Scalability: build for future growth, for instance in order to allow other media like video to be introduced into the solution
- Applying best practices in designing, setting up and deploying the data network used for voice services

To consistently ensure the highest quality of voice connections, the following points are recommendations according to industry best practices when implementing IP Telephony. Note that these suggestions are only options and might not fit individual business needs in all cases.

- <u>Quality of Service (QoS)</u>
  QoS for real-time voice packets on a data network is obtained by tagging voice packets as having priority over data packets. Networks with periods of congestion can still provide excellent voice quality when using a QoS policy. The recommendation for switched networks (using switches in a Local Area Network LAN) is to use IEEE standard 802.1p/Q.
  The recommendation for routed networks (using Routers to interconnect multiple networks) is to use DiffServ Code Points (DSCP). The recommendation for mixed networks is to use both.
  Port priority can also be used to enhance DiffServ and IEEE 802.1p/Q. Even networks with sufficient bandwidth should implement CoS/QoS to protect voice communications from periods of unusual congestion that a computer virus might cause.

- <u>Switched network</u>
  A fully switched LAN network is a network that allows full duplex and full endpoint bandwidth for every endpoint that exists on that LAN. Although IP Telephony systems can work in a shared or hub-based LAN, best practice recommendation is to exclusively use a switched network for IP Telephony for

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consistent high performance and optimum voice quality.

- Network assessment
  A Basic Network Readiness Assessment is vital to a successful implementation of IP Telephony products and

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Appendix B Short Excursion into Erlang's Traffic Theory

Appendix B *Short Excursion into Erlang's Traffic Theory*

In general, Erlang said that the wider a channel is the more traffic it is capable to carry (not a surprise).

Erlang (Erl) is a measure for traffic capacity carried by a system. In case we have a single line with a theoretic capacity of 1 Erl this means that this line is occupied 100%, so 60 minutes within 1 hour while we watch this line's occupation. This can achieved in two ways: Having a single call with a duration of 60 minutes, or having multiple calls immediately back to back during that one hour. Do we agree that both are somewhat unrealistic and non-practical?

In reality we have calls with a statiscally distributed arrival and call length. So probably we have a call being picked up by the call taker and occupying him for 3 minutes. During these 3 minutes, we have another call arriving, which he can't take, so the call is lost. During our 1-hour window of observation we have seen no other calls arriving. In this case, the traffic capacity is 3 min out of 60 min which equals 5% or 0,05 Erl.

If he would have been able to take this call 5 minutes later, this would have been a 17-minute call. So in that case both calls would sum up to 20 minutes, which equals 33,3% or 0,33 Erl.

In case we had another call taker available to pick up the call while the first call taker was busy, we would have come to a result or 20 minutes out of 2 x 60 = 120 minutes, which leads us a capacity of 16,7% or 0,16 Erl.

So we see that widening the access channel helps raising

All all of the Erlang calculation is based on statistically validated models, and the Erlang values for systems can be picked from Erlang tables or computed with special algorithms, following the famous Erlang B formula :-)

$$P(c) = \frac{\frac{\rho^c}{c!}}{\sum_{i=0,c} \frac{\rho^i}{i!}}, \text{ where } \rho = \frac{\lambda}{\mu}$$

When it comes to very small entities, e. g. a single line, we see that they are truly inefficient.

Let's review an example which could be appropriate to emergency services:

According to Erlang B, a communication system equipped with 5 lines (equals 5 call takers) is capable of carrying 1,36 Erl, with a blocking probability of 1%. Translated into a more operational language this means that assuming a maximum of 1% of all offered calls being blocked, 5 call takes are statistically able to carry 1,36 calls in parallel on an average.

So every PSAP with only 5 call takers is falling into that category, and if we have 5 of those PSAPs disconnected from each other and working as "islands", they all together would capable of carrying

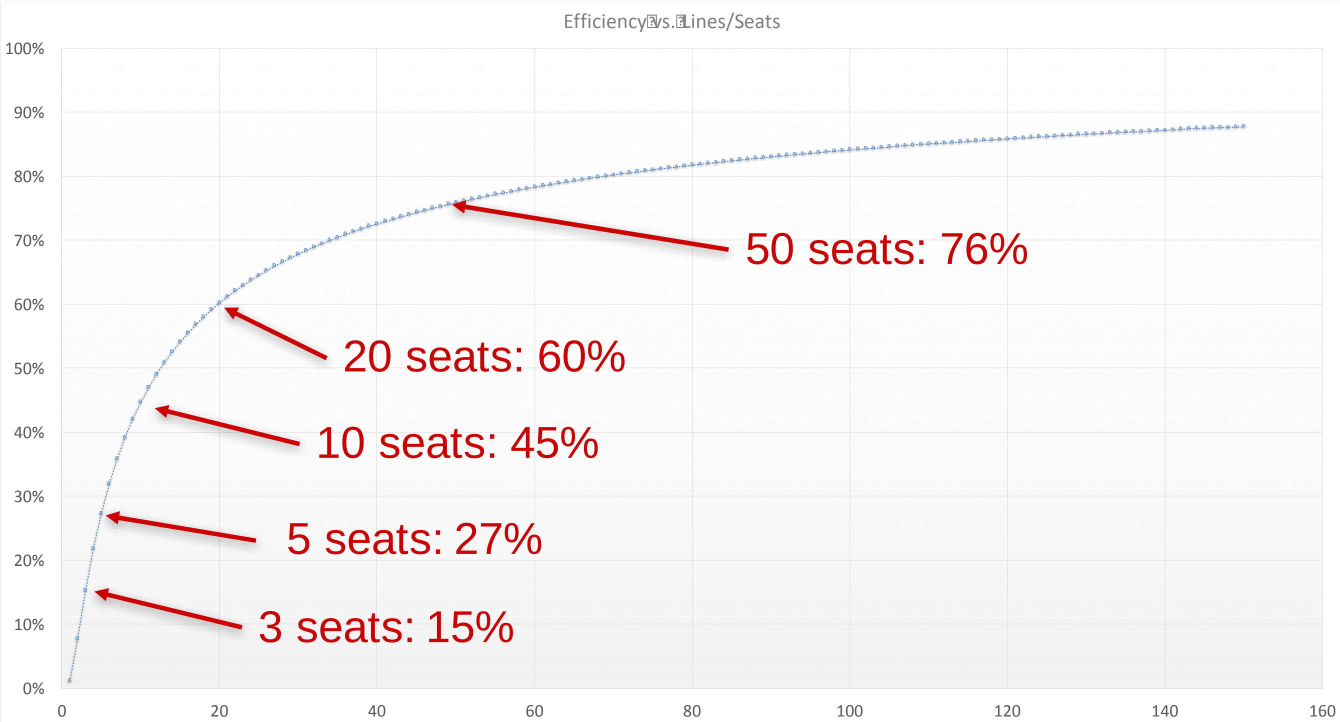
5 x 1,36 Erl = 6,8 Erl

of traffic.

Now, if we would connect these 5 PSAPs and make them act as one, we would come up to a community of 25 call takers, and for 25 call takers Erlang B does return a value of

16,1 Erl

of traffic, which is a factor of 2,37 just by joining forces. The graphic below shows the traffic efficiency as a function of available lines respectively call taker seats:



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