Stratix

REPORT

Digitisation of wireless microphones
The effects on spectrum use

Management Summary

The PMSE (Programme Making and Special Events) -sector is facing a difficulty with the number of wireless devices that are deployed during events and the amount of radio spectrum available to these devices. Intermodulation between transmitters used to result in needing more spectrum per device with each extra transmitter added to an event. This would lead to a spectrum crunch where more spectrum was needed than was available for PMSE-events. This report has found that this problem to a significant extent is solved through higher quality and more expensive microphones and in-ear monitors (IEMs) that make use of isolators/circulators to filter out any intermodulation products from retransmission of other devices. It doesn't matter whether these devices are digital or analogue as both can yield similar gains over traditional devices that don't have good filtering of these intermodulation products.

The Dutch Radiocommunications Agency had two main questions for this study:

a) What type of PMSE applications that make use of the UHF band can be digitised and what types of applications cannot?

b) How much spectral savings could be obtained from digitisation of the applications that can be digitised, given the demands and constraints at large media event?

The implicit assumption was that there would be strong benefits if applications could be digitised. Particularly digitisation would allow the more efficient use of spectrum, because it would be less sensitive to intermodulation.

Retransmissions of the signals of other equipment due to intermodulation are particularly common in PMSE as many transmitters and receivers share a physically small space and pick up and retransmit each other's signal. What the study found is that intermodulation isn't solved by digital technology per se. Intermodulation is solved by equipment that is less sensitive to intermodulation products. This is achieved by using isolators/circulators. All digital PMSE-equipment comes with isolators/circulators. Low-end analogue equipment may not have isolators/circulators on board to keep costs low. High-end analogue equipment does come with isolators/circulators and as a result do not suffer from intermodulation products either. When comparing high-end digital microphones with high-end analogue microphones it appears there is no fundamental difference in their spectrum efficiency. Any observed differences appear to be more manufacturer related, then fundamentally caused by the underlying characteristics of analogue or digital transmissions.

The strict answers to the questions of the Radiocommunications Agency are that only microphones can be digitised given the constraints of the PMSE-sector. IEMs, though they could deliver digitised sound, are less likely to be digitised. The sound-engineer may prefer digital microphones as they are likely to deliver a more "clean" and consistent sound. Though analogue microphones could deliver similar quality, they may not always be as clean and consistent, because of the use of companders, which have less straight forward artefacts compared to digital. It is hard to keep digital microphones under 2-5ms latency. As the total latency in the chain from microphone to IEM shouldn't exceed 5ms this means that IEMs will remain analogue and the chain will be a mix of digital and analogue transmitters.

The move to higher quality microphones that use isolators/circulators is likely to allow between 16 and 23 microphones and IEMs to be planned in an 8MHz band. This will aid medium scale productions more than large scale productions. Medium scale productions will have to take IM-products less into consideration and plan around 60 devices without the aid of a dedicated spectrum planner. Larger scale events however already have a spectrum planner, they also make use of higher-end equipment. The combination of the two means that higher-end events are already operating at around 16 (or more) devices per 8MHz and likely more if special separation etc. are available to them. If high-end equipment is used, many events and locations will be able to handle even uncommon events provided that spectrum planning is done right. With help from the Radiocommunications Agency even exceptional events such as a crowning, a Eurovision final or a European football championship are manageable. Large scale productions will find themselves in difficulty sometimes managing all the equipment and planning it correctly, because spectrum can be limited and unforeseen other secondary use is not always easy to mitigate. This is not unmanageable, however it will require more effort and therefore investment from the event organisers.

Contents

1   Introduction

The programme-making and special events (PMSE)-sector uses radio spectrum to transmit audio (microphones/headphones), video (cameras) and other media or command/control information wirelessly. The applications that locally use radio spectrum can be as small as a single microphone for a speaker at a meeting or as large as 400 wireless links at musical theatre, sporting event or the recording/broadcast of a television show. Traditionally the PMSE sector acts as a secondary user of the UHF-spectrum (e.g. 470-862 MHz) that was allocated for television broadcasts, utilising the white spaces and gaps between channels. Technological developments, such as digitisation and mobile communications have changed the spectrum landscape such that the traditional bands the PMSE sector used are not unused anymore and do not allow secondary use. This limits larger scale events in the number of links and the quality of the recording they can use.

1.1 Research questions

The Radiocommunications Agency Netherlands has in cooperation with the industry representative PMSE.nl indexed what kind of constraints there are for the availability of spectrum for wireless microphones if in the near future the 700 MHz band is used for mobile networks. These constraints are likely to appear at large media events. This research will therefore focus on large media events.

However the basis for the analysis of constraints for large media events is based on conventional (analogue transmission) technology, where the possible benefits of digitisation weren't taken into account. The most important expected advantage of digitisation of wireless microphones is the more efficient use of spectrum. This innovation appears necessary to remove the constraints.

There are a number of different kinds of audio PMSE applications that make use of the UHF broadcasting band. Because digital systems incur (additional) latency in the transfer of signal there are likely to be problems in the digitisation of certain time-critical applications.

The first question to be answered is:

1. *What type of PMSE applications that make use of the UHF band can be digitised and what types of applications cannot.*

In order to answer these questions it will be necessary to define what can be considered 'digitisation' and to research for each type of PMSE application, for example registration and the In-Ear Monitors used by singers and musicians, to what extent the introduced limitations as a result of digitisation are still acceptable and workable, or the introduced advantages of digitisation can be utilised, such that an objective overview can be obtained.

The additional question that should be answered is:

2. *How much spectral savings could be obtained from digitisation of the applications that can be digitised, given the demands and constraints at large media events*

To answer this question it is necessary to research a number of PMSE scenarios of large scale events in two different ways. One where the conventional audio PMSE situation is researched and the other (where part of the) audio PMSE devices (those that were identified in the first question) have been digitised

To paraphrase the questions: Would it be possible to digitise PMSE audio used at large events to save spectrum or hold more devices in the same spectrum? When this question is answered it is possible to quantify the total audio PMSE spectrum need. On this basis, but outside the scope of this report, it is then possible to determine whether there is sufficient spectrum after the 700MHz has been reassigned and new bands have been added in accordance with international regulations.

1.2 Research method

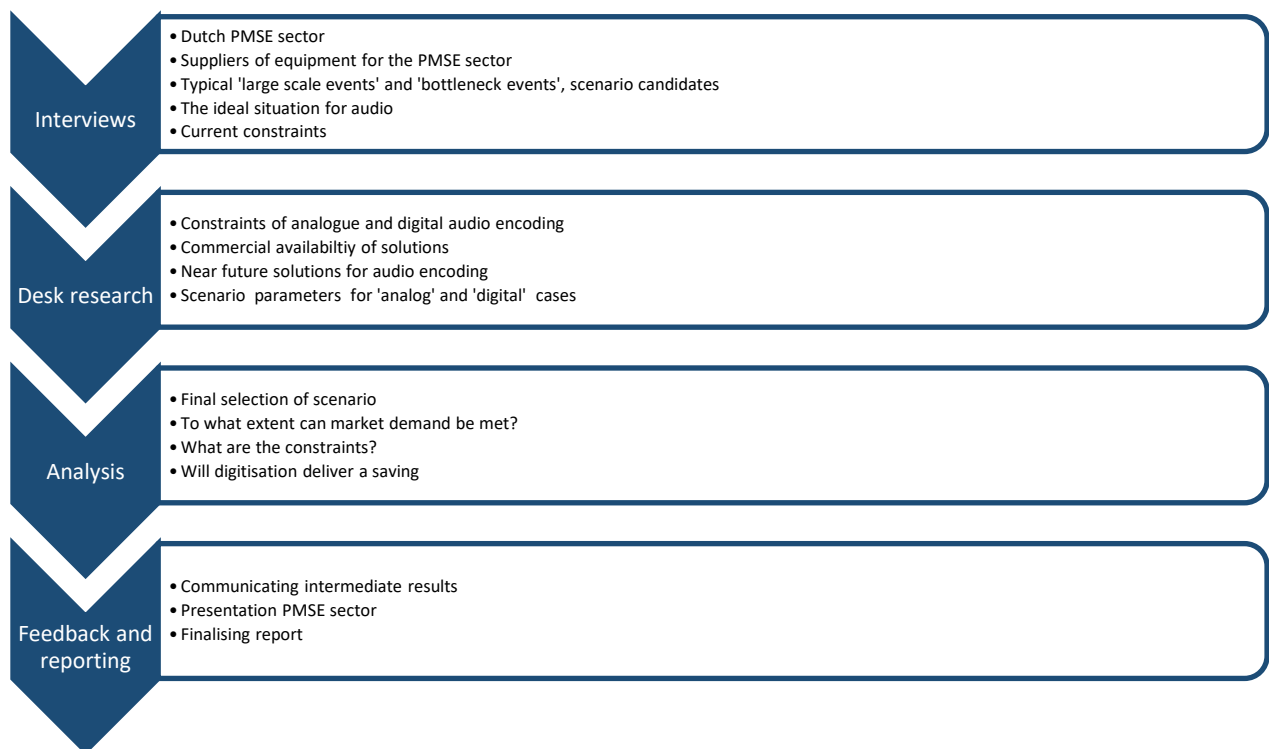**Figure 1:** Process steps

To determine the need of the PMSE sector and large events Stratix interviewed large event audio specialists, manufacturers and researchers on their ideal situation with regards to audio. The interviews were used to determine the scenario elements for 'typical large scale events' or 'bottleneck events', e.g. events with the highest or most difficult demand for radio spectrum.

Interview topics were also the current limitations and how this affects the production of large events. This set a baseline for their needs and what constraints and trade-offs they currently face with regard to audio quality, number of sources, interference, transmitter spectral mask etc.

In parallel with the interviews we have executed desk research into the theory and reality of audio encoding for large scale events. This included research on digital audio encoding geared towards encoding for broadcast, playback or communication, near real-time encoding, mixing and playback. In addition also work on analogue compression and transmission, impact of different compression and transmission techniques, on intermodulation products and robustness of technologies with regard to possible interference effects of intermodulation products were considered in this context.

In the analysis our team considered the most strenuous scenarios for different types of events, and looked for the implications and possible trade-offs when looking at the future of audio and digitisation in the PMSE-sector. Finally conclusions were drawn regarding the benefits and drawbacks of digitisation over analogue audio in large scale PMSE events and recommendations were formulated.

1.3 Scope and definition of digitisation

During the research it became clear that the term digitisation is by itself not very clear. In the case of PMSE it could mean the digitisation of the audio signal or the digitisation of the transmission. When evaluating the benefits however it is necessary to know what part is digitised. In this report we therefore take great care to evaluate the benefits and drawbacks of the digitisation of the audio signal separate from the benefits and drawbacks of the digitisation of the transmission. Chapter 4 will focus on digitising the audio. Chapter 5 focuses on the digitisation of the transmission.

1.4 Reader's guide

In chapter 2 the background and context of the PMSE sector and the use of wireless technologies is briefly described. Also an overview of the current situation and trends in this sector is given with regards to allocated spectrum and numbers of devices used per event type. In chapter 3 technical considerations that surround PMSE-sector use of wireless audio are discussed. For example the need for low latency and good battery performance. Chapter 4 discusses the information theoretical aspects of digital audio encoding and particularly why digital is harder given the PMSE-requirements. Chapter 5 then evaluates the differences between digital and analogue transmissions for PMSE-applications. Chapter 6 evaluates some practical scenarios for the PMSE-sector, where digital and analogue audio are compared. And chapter 7 gives the conclusion and recommendations.

Early on it was recognised that the part of the PMSE-sector most affected by a possible spectrum scarcity are large scale events where a large number of microphones and in-ear monitors (IEMs) are used. Concerts and festivals are particularly demanding as they require

high quality audio and large number or devices. Sports events and large scale news events can be demanding in the number of devices. It is for this reason that this report uses these events as its primary examples. Other uses are important too, but generally will be less affected by spectrum scarcity.

2 Background

This paragraph describes the background behind the question on spectral efficiency of PMSE. It describes what the PMSE sector is and its importance to society. How wireless technologies are used in PMSE. The scale of the events and how this puts constraints on wireless and how this environment is changing.

2.1 The PMSE-sector

The PMSE sector is a very diverse and fragmented sector. Under the term PMSE fall concerts, festivals, (musical) theatre shows, movie-shoots, fairs, sports events and live and recorded television formats. The focus of this report is on large scale use of wireless communication, for microphones, in-ear monitoring and intercom. Therefore for this report the exact distinctions between the various forms of PMSE is less relevant, however we will give some insight into the diverging needs of the various parts of the PMSE-sector.

The size of the PMSE sector should not be underestimated. Every year a variety of events have over 56 million attendants and over 2 billion in revenue. There are over 800 festivals each year in the Netherlands, with over 23 million attendants with roughly 750 million euro in revenue.[1] An estimated 1 million tickets are sold to major music concerts, with close to 100 million in revenue.[2] The Dutch Central Bureau for Statistics (CBS) estimated 18.1 million attendants to 52.5 thousand performances in 334 Dutch theatres and music venues with over €750 million in revenue.[3] ING estimated that the Dutch electronic dance sector adds €200 million to Dutch exports. There are 170 dance events per year, attended by over 2.7 million people with roughly €200 million in revenue.[4] Fairs and tradeshows have 6.8 million visitors, with 400 million turnover.[5] In addition to direct revenues there are additional revenues from hotel visits, food and drink outside the venues etc. The two largest commercial television broadcasters and the public broadcasters have a combined revenue of €1.5 billion in television. There is likely to be some overlap in these numbers, but it is clear that the PMSE sector is quite significant to the Dutch economy. PMSE plays an important role in the lives of many Dutch people, as part of media productions, leisure and recreation activities that are consumed in or outside their homes. The use of wireless in PMSE

Amplified and recorded sound (and imagery) is essential to PMSE. Even if the audience can see what is happening, hearing it is not easy for anyone more than a few meters away. This

---

[1] http://www.vvem.nl/nieuws/110-berichten-van-derden/1054-festivalsector-trekt-recordaantal-bezoeken-in-2015

[2] https://decorrespondent.nl/5136/zo-werd-live-nation-het-machtigste-bedrijf-van-de-muziekindustrie/612026930592-e26c2274

[3] Bezoek professionele podiumkunsten kruipt uit het dal 10-1-2017 https://www.cbs.nl/nl-nl/nieuws/2017/02/bezoek-professionele-podiumkunsten-kruipt-uit-het-dal

[4] Nederlandse dance-industrie van toenemend economisch belang, ING, 27-05-2015, https://www.ing.nl/nieuws/nieuws_en_persberichten/2015/05/nederlandse_dance_industrie_van_toenemend_economisch_belang.html

[5] Factsheet Beurzen 2015 , Respons http://www.respons.nl/uploads/Factsheet-Beurzen-2016.pdf

has been true throughout time with amphitheatres, concert halls and theatres which were designed with acoustic properties in mind[6]. Electronic recording, transmission, amplification and communication made it easier to reach the furthest parts of the event location, to record, distribute and coordinate the event. Microphones, speakers, in-ear monitors and intercom are therefore essential.

Initially everything was wired and powered of the electric grid as batteries and radio technology weren't strong and miniaturised enough. This restricted artists and staff in their movements and performances. At first there was only one wired microphone in a studio or on stage. This microphone was for the singer, whose voice was the least powerful instrument. With advances in microphones, speakers and mixing consoles the possibility evolved to mix together multiple instruments. In the early 1950s the first wireless microphones were invented for use in sporting events and performances. Their range was initially as little as 4.5 metres[7].

Wireless microphones initially were a niche; they weren't powerful enough, didn't offer high enough audio quality, too heavy, too little battery life. However microelectronics facilitating miniaturisation made it possible for performers, actors, sportsmen and referees to carry or wear the equipment. Wireless microphones at first made use of VHF spectrum (where radio also resides), however in these bands they suffered from interference of other radio sources and limited spectrum availability. Over time those that had to be used in high quality situations moved to the UHF spectrum in the white space between UHF television channels. This spectrum was available because analogue broadcasts suffered from interference if channels were spaced too close together and therefore some space between the channels was allocated. Channels couldn't overlap between neighbouring regions either as they would interfere. In addition there weren't as many channels active as technically possible. For example the Netherlands had at first only one, later two and even more later three broadcast channels active. As the PMSE sector in part originated from the broadcasting industry it was natural for them to regard that spectrum as available for use for auxiliary purposes related to the broadcasting industry, such as wireless microphones and other wireless links.

The applications of wireless microphones and transmission also moved towards instruments, for example to allow electric guitars to be used wirelessly, increasing the movement of the guitarist around stage. For other instruments one or more wireless microphones and transmitters are used to transmit the sound of the instrument. Multiple microphones are needed to capture low and high frequencies. Multiple transmitters can be used on for example accordions to allow the hands to move freely. Just as society followed a trend towards wireless and miniaturisation, so did the PMSE-sector with increasing numbers of wireless devices. In some areas, such as the Hilversum Mediapark studio village, where most of Dutch broadcasting is located this growth became exponential in the 2000-2010 (see Figure 2, which is based on data collected by CEPT from the Mediapark)

---

[6] Akoestiek van de 'schoenendoos', NRC 04-03-2014, https://www.concertgebouw.nl/uw-bezoek/gebouw-geschiedenis/beroemde-akoestiek/nrc-handelsblad-akoestiek-van-de-schoenendoos
[7] Company history of Shure, http://www.shure.com/americas/about-shure/history
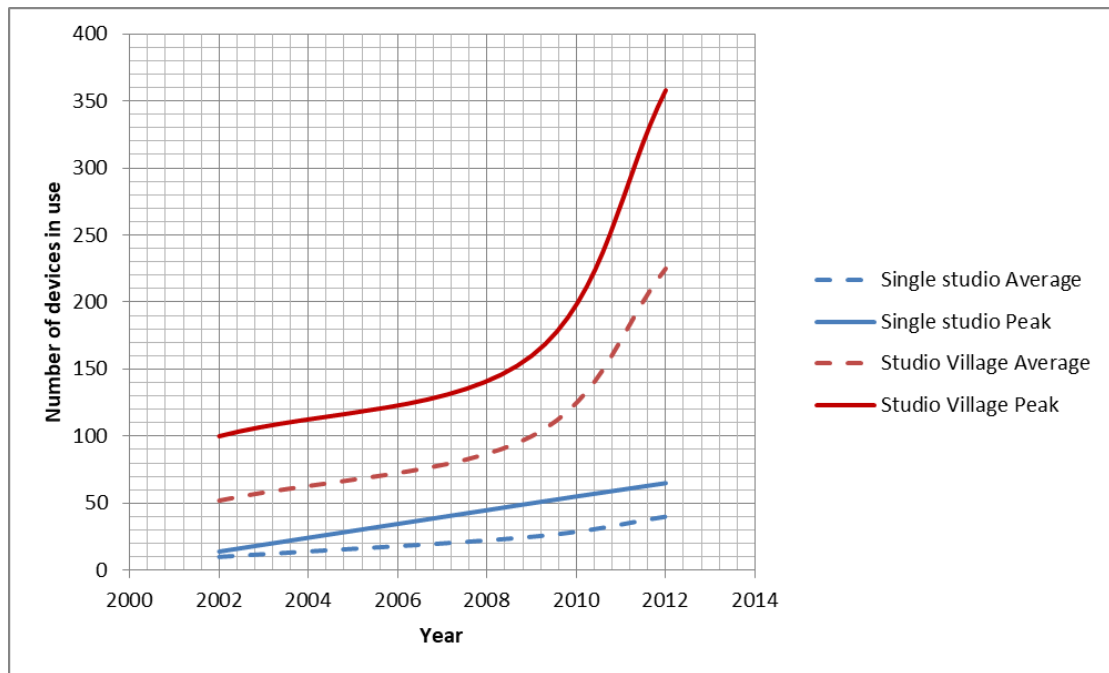
Figure 2: Projected growth of number of devices in studios (source: CEPT[1] graph Stratix)

In programme making and movie making wireless microphones made it possible to have a microphone per actor instead of using the large booms that were hanging overhead of the actors. These booms would restrict the freedom of the cameraman and as a result the flexibility and artistic freedom of the director. In electronic news gathering wireless allowed the reduction of crews to a combined camera/sound operator and presenter and more recently even one person crews. These can then walk around events, concerts and conventions to do interviews more flexibly then before. At larger scale programmes, such as the Voice (of Holland), technicians now deal with 100 to 200 wireless links, for artists, band, presenters, judges, staff etc. Not only do they need microphones for singers and instruments, but also in-ear monitoring and intercom functions.

Today wireless microphones can be found everywhere. Hotels, conference centres, museums, hospitals, universities all employ them in some form or other to allow talks, lectures, instructions and communication. Though the focus is primarily on large scale PMSE, these other uses cannot be excluded from the overall picture.

Digitisation of wireless microphones is relatively new on the market. The first digital microphones were introduced in the market around 2006. The major brands such as Sennheiser and Shure took until 2011-12 to bring digital wireless microphones on the market. To bring this into perspective, the GSM digital telephony system was brought on the market in 1991. Early digital PMSE systems had relatively high latency of over 4 ms, which is considered too long for professional performances. Only when professional systems offered lower latencies, were they considered for large scale professional performances, which are the topic of this research. Low latency systems that are good enough for professional performances are only available since a few years. The expected benefits of digitisation lie in a better sound quality and supposedly less susceptibility to interference from intermodulation products. To determine in what respect this is actually the case is the main reason for this

research. In-ear monitors are also a recent development. Though some development started in 1987, significant commercial development didn't start until ten years later. Monitor speakers also known as wedges were initially developed in the 1960s to help artists, such as the Beatles, hear what they were playing over the screaming of their fans. For some bands this meant that the speakers had to be very loud, which caused problems for the hearing of the artists and feedback issues with the microphones picking up the sound of the wedges and different sound quality and levels on different parts of the stage. In-ear monitors allowed the wedges to be removed and the musicians to insulate themselves from the sounds of the audience. In addition each artist can receive their own personal mix, for example a singer who needs to tune to the piano could receive more piano, whereas a drummer would like to hear more of the bass player and vice versa. Wireless was essential as musicians wanted the freedom to move across stage. Developments from personal in-ear audio, such as the iPod drove miniaturisation and brought costs down from initially several thousand to several hundred euros, enabling more and more artists to make use of IEMs.

In addition to the talent that is the focus of the performance, there are a great number of people working to operate sound, light, special effects, security and all kinds of other things that happen on and off stage. For these people intercom systems are necessary. Though some will not move around, such as the technicians at monitor consoles or the light operators, others do have to move and will use wireless intercom systems to communicate. These systems do add to the number of wireless connections needed at events. These systems used to make use of the same technology and frequencies as wireless microphones and IEMs, but are increasingly moving to other technologies, such as DECT. This transition is relatively recent and technicians are debating to what extent these systems are usable, but there is agreement that the intercom will move out of the PMSE bands.

Miniaturised wireless microphones and IEMs caught on so well that today at music festivals almost all artists play with them, resulting in over 300 active parallel wireless links. At Pinkpop 2016 Paul McCartney was reported to be the only artists not to use IEMs. At an event such as Glastonbury in the United Kingdom over 1400 links for artists, journalists and intercom are necessary. Some in the industry report that each year the number of wireless links active at such events increases by up to 10%. This would result in a doubling every 7 years. (Figure 3)
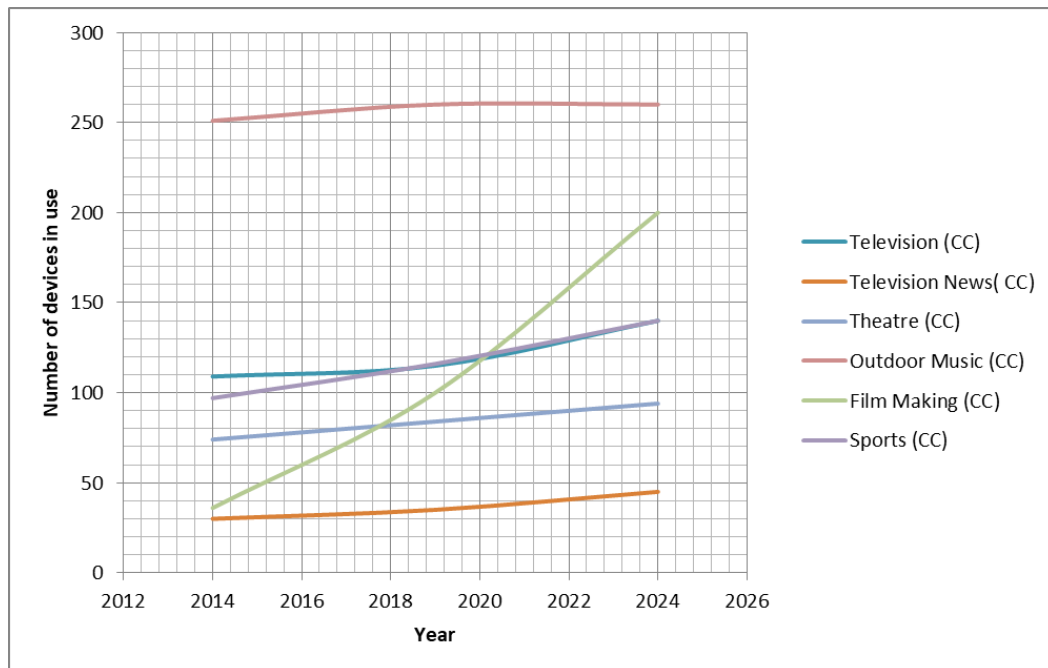
Figure 3: Estimates and predictions for wireless devices per event type.
Source: Cambridge Consultants [2]

2.2 The scale of an audio event

Spectrum planning for audio events is considered by the industry to be a necessary evil. In principle they want a plug and play experience. However the expansion in the number of devices and the decrease in the available spectrum have created a crunch, where serious thought and planning goes into planning larger scale events. In addition to the scale, the location of audio event and the unlicensed use of other users than the PMSE sector are possible and can create additional problems. A humorous example of the impact of other users, that impacts larger scale PMSE use, that was given was the opening of a gym at the Hilversum Mediapark in 2014. This gym made use of wireless audio for its gym instructors to give classes. The effect was that the studios across the Mediapark had to stop using those frequencies for television productions. Though this is a single incident, the use of wireless in a conference or in a hotel can affect a nearby event.

Table 1 shows a classification of the types of events and their use of spectrum, developed by Anker and Pierens[8].

Table 1: classification of PMSE event types (Source: Anker & Pierens)

Level	Equipment	Use in NL	Density
A1 end-user	1-3 mics	Always	<12 /km^2
A2 starting eng.	4-8 mics 2 in-ears	Each day >30 studios Numerous events	<24/km^2

---

[8] Developed informally by Peter Anker of the Dutch Ministry of Economic Affairs and Eric Pierens of NEP. This appears to be its first appearance in a report.

B1 sound engineer	4-12 mics, 2-4 in ears 2x duplex coms 2x simplex coms (walkie talkie)	Each day/multiple TV, Theatre' Trade show	<50/km^2
B2 exp sound eng	12-24 mics 12 in-ears 3x simplex Wireless cam Wireless data Electronic News Gathering Other systems (sponsors, presentation etc)	3-7 days/week Large musical Large concert Large TV-show Event/trade show	<100/km^2
C1 freq planner	24-150 mics. 12-30 in-ears 30 duplex 14 simplex Wireless cams, data, ENG	Every weekend Mediapark Large trade show Music festival Parliament University	>100/km2
C2 exp freq planner	>150 mics >30 in-ear > 30 duplex > 30 Simplex ~16 camera's Helicopters, motor cycles point to point etc.	Once/Twice a month International event European championship Music awards Major music festival	>100km^2 Many specials

What this classification illustrates is that there is a wide variety in demand for wireless audio. Large scale events happen weekly in a country like the Netherlands, sometimes multiple times in a week or weekend. Even the extra-ordinary events can be as regular as twice a month (particularly in summer). Such extra-ordinary events are large scale music festivals, such as Pinkpop, Lowlands and Zwarte Cross, but also large sport events such as European Championships, Amstel Gold, Eneco Tour, etc.

2.3 The changing environment of spectrum

The purpose of this report is to evaluate the effects of wireless digital audio on spectrum use. The rationale behind this is that large scale events state they are running out of wireless spectrum, because the number of wireless link is increasing each year, whereas the available spectrum is decreasing. The spectral mask of a PMSE-link is limited to 200 kHz. Planning links in an 8MHz channel however requires a certain space between links in the spectrum. Interference (from other sources, but also through intermodulation, where transmitters pick-up and retransmit each other's signal) greatly reduces spectral efficiency, so that between 10 and 16 devices can safely be planned in an 8MHz band.

Currently there are a number of bands available (see Table 2). However, the PMSE sector is a secondary user, where Digital Television or radio is the primary user. The effect of this is that in all locations in the Netherlands at least 5 times 8 MHz or roughly 60-80 microphones have to be deducted from row E, because this is in use by DTV. In addition in areas where there is an overlap between DTV regions, 10 and sometimes more channels have to be

deducted. This is even more so where the location is close to the German and or Belgian border, where DTV channels from these countries occupy channels too (2 for Belgium and up to 7 for Germany). This spectrum might be usable, but it would require more care and knowledge of the technician using the spectrum.

Furthermore all the VHF- channels (rows A to D) should be considered difficult for use by high quality audio. Typical VHF suffers from a lot of interference e.g. from electronics and LED light walls. CEPT cites as a concern that low frequencies require large antennas. The noise floor and clock frequencies in electronic equipment may create interference to audio PMSE applications. It is for this reason most wireless equipment makes use of UHF. So effectively for the PMSE sector 470 to 791 MHz minus at least 40 MHz is currently available, though 694 to 791 MHz will be lost too after 2020 to mobile communication. It is likely new spectrum becomes available, but which spectrum and how much is currently uncertain. Propagation characteristics and body absorption vary from band to band. Therefore spectrum slices of equal size but at different bands are not fully identical. Doubling frequency typically costs a 7-10 dB hit in link budget depending on environment. Unfortunately due to intermodulation effects, which will be discussed in further sections, not all spectrum can be used (depending on the equipment used).

Table 2 : Frequencies available in The Netherlands for PMSE (Source: Dutch RA)

	Frequency (MHz)	Power
A	36,600 – 36,800 37,000 – 37,200 37,800 – 38,000 38,200 – 38,400 38,600 – 38,800	10 mW e.r.p.
B	37,480 - 37,600	10 mW e.r.p.
C	97,5 – 108	50 nW e.r.p
D	195-202	50 mW e.r.p.
E	470 – 556 558 – 564 566 – 572 574 – 580 582 – 588 590 – 596 598 – 604 614 – 694 694- 791[5]	50 mW e.r.p.
F	823 – 826	20 mW e.i.r.p. voor handheld apparatuur 100 mW e.i.r.p. voor apparatuur die op het lichaam wordt gedragen
G	826 – 832	100 mW e.i.r.p.
H	863 -865	10 mW e.i.r.p.

	Frequency (MHz)	Power
I	1785-1805	20 mW e.i.r.p. voor handheld apparatuur
		50 mW e.i.r.p. voor apparatuur die op het lichaam wordt gedragen

In order to help the industry do frequency planning the Dutch Radiocommunications Agency has a spectrum planning tool available online[9]. It gives conservative estimates of available spectrum for outside and indoor use. The table below shows the difference by location for outdoor use at 4 major PMSE locations and the effect of the 700 MHz band to be used for cellular communication. What becomes clear is that after 2020 the Pinkpop festival will be operating in 3 bands, or even less, as there is likely one channel in the Netherlands and 1-3 channels in Germany that need to move to a lower channel. For each of the locations the yellow part are the bands that will be used by cellular and the bottom row lists how much spectrum will be lost and also mentions the number of Dutch DTV channels that need to be relocated. Given that 10-16 devices per 8MHz block is generally considered the industry standard today, this would limit an event like Pinkpop to 30-50 wireless links.

Table 3 : Spectrum availability outdoor for major event locations

	Pinkpop Festival, Landgraaf		Mediapark, Hilversum		Amsterdam Arena		Lowlands, Biddinghuizen	
Available Frequency bands Marked yellow: (to be) used by cellular	526-534	MHz	470-478	MHz	470-478	MHz	470-478	MHz
	542-550	MHz	478-486	MHz	478-486	MHz	502-510	MHz
	622-630	MHz	502-510	MHz	502-510	MHz	534-542	MHz
	758-766	MHz	534-542	MHz	526-534	MHz	542-550	MHz
	790-791	MHz	542-550	MHz	534-542	MHz	550-556	MHz
	823-830	MHz	550-556	MHz	542-550	MHz	582-588	MHz
	830-832	MHz	558-564	MHz	550-556	MHz	598-604	MHz
			566-572	MHz	558-564	MHz	622-630	MHz
			598-604	MHz	566-572	MHz	630-638	MHz
			622-630	MHz	598-604	MHz	646-654	MHz
			630-638	MHz	622-630	MHz	670-678	MHz
			646-654	MHz	630-638	MHz	686-694	MHz
			670-678	MHz	646-654	MHz	710-718	MHz
			686-694	MHz	670-678	MHz	718-726	MHz
			710-718	MHz	686-694	MHz	734-742	MHz
			734-742	MHz	710-718	MHz	750-758	MHz
			742-750	MHz	734-742	MHz	774-782	MHz
			774-782	MHz	742-750	MHz	790-791	MHz
			790-791	MHz	750-758	MHz	823-830	MHz
			823-830	MHz	774-782	MHz	830-832 MHz	
			830-832	MHz	782-790	MHz		
					790-791	MHz		
					823-830	MHz		
					830-832 MHz			
Netto available:	**42 MHz minus 9MHz + 1**		**146 MHz minus 33 MHz**		**170 MHz minus 49 MHz**		**140 MHz minus 41 MHz +**	

[9] Microfoonbanden, https://www.agentschaptelecom.nl/microfoonbanden/

DTV channel = 17MHz	+ 2DTV channels = 49MHz	+2DTV channels 65MHz	2 DTV channels = 57MHz
	=		=

A more practical view is to use a spectrum analyser. Two scans of the 2017 situation at Lowlands and Pinkpop show the situation at two main event locations (Figure 4 and 5). The Lowlands scan shows the DTV channels that are used and some noise from DTV channels from another transmitter that is further away, with 3 bands of LTE on the right side. What is clearly visible is that the Pinkpop spectrum is much more noisy and the DTV channels of the Belgium and Germany can be seen quite well, whereas on the right side the LTE-bands that are used for 4G stick out.

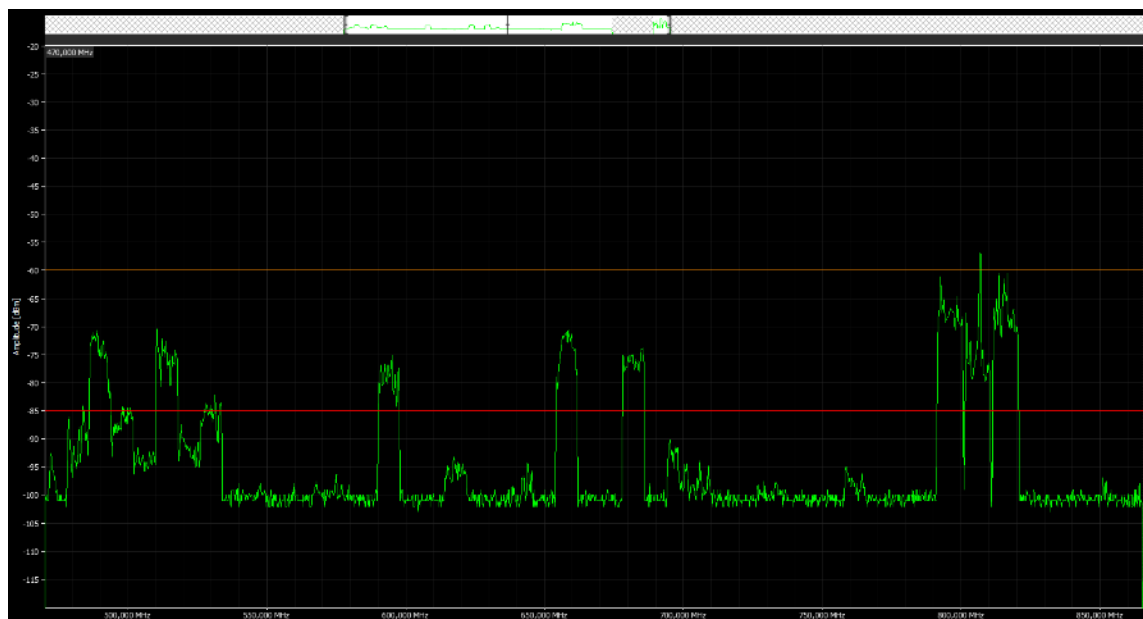**Figure 4** Lowlands Biddinghuizen 2017 (c) Kees Heegstra, Camel & co
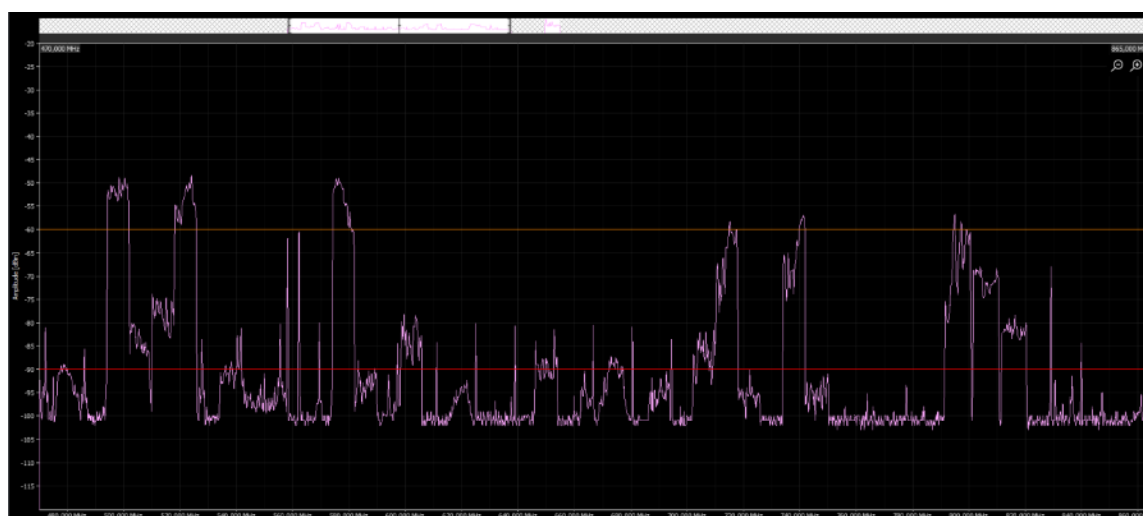

Figure 5 Pinkpop Landgraaf 2017 (c) Kees Heegstra, Camel & co

There has been discussion of using other bands for wireless microphones. However so far there hasn't been a permanent solution at the EU. For smaller scale applications the 2.4 GHz band is sometimes used, however it is the band that is also used for WiFi and a number of

other applications, so it can't be used too intensively as this undoubtedly leads to clashes with other uses, such as ad-hoc WiFi networks and Bluetooth.

Discussions with industry have shown that the professional users of spectrum for wireless microphones see the bands listed by the Radiocommunications Agency as a guideline. As one of them said: In the end it is about the noise floor and how high you can go above it, without causing trouble for others and how low you can stay without causing trouble for yourself. Such sentiments are resonated internationally by other experts.[10] This means in practice that C-level planners will be able to plan a larger number of devices across bands that are not on this list. A and B-level planners will stick more closely to the bands designated by the Agency.

---

[10] An in detail examination of the concerns that spectrum planners for large events have is described by Steve Caldwell when he describes the planning necessary for the opening of the Rio Olympics 2016. http://soundhub.audio/rf-coordination-rio-2016-steve-caldwell/ and a more standard example in Australia http://www.jands.com.au/support/product-support/audio-technical-materials/wireless-spectrum/the-great-wireless-spectrum-shortage-myth-or-reality

3 Requirements for PMSE-audio

The first question to be answered is:

1. **What type of PMSE applications that make use of the UHF band can be digitised and what types of applications cannot.**

This questioned is answered by first evaluating under what demands users put on a PMSE-application. Parameters, such as sound quality, latency and cost are important from a user's perspective. From a policy perspective spectrum use is important. Types of applications evaluated are microphones and the in-ear monitors used by singers and musicians. This chapter looks at the requirements that users have for the PMSE applications. The following chapters evaluate whether digitisation of either the sound and/or the transmission can deliver benefits in achieving those requirements.

The requirements that can be derived from main components of a PMSE system: Production, Archive and Distribution. It is concluded that PMSE-audio needs a high availability system, with low latency, low energy consumption and high quality.

3.1 The main components of a wireless PMSE system

A wireless PMSE-system consists of two main parts. The first part is the way the audio is encoded. The second part is how it is transmitted. Encoding focuses on capturing as much as possible of the audio and representing it either in a waveform or a string of bits. The second part is the wireless transmission part. This takes all of the waveform or bits presented to it and then in the case of digital adds data (redundancy, checksums, channel sounding, training, synchronisation, signalling) so that the receiver knows what should be received and whether it was received correctly. The transmitter then maps the bit stream to a symbol stream and converts this so to create an electrical signal to carry the data. When spectral savings are important the first question is, whether, in order to capture a similar input, digital systems need more, less or equal "spectral footprint" compared to analogue. If for example compression would allow a digital system to transmit less while achieving the same and/or higher audio quality this would translate through to how much spectrum would be needed.

The second part of the system might yield benefits if the transmitted signal would need less spectrum to be transmitted using a digital system. Or, if it would require more, whether the combination of savings in encoding would mitigate the increase in spectrum use. It is therefore important for this research to look at these two elements. In communications encoding/compression is named "source coding" and protection against transmission errors is named "channel coding". Aside of source and channel coding there is also the challenge of spectral mask, which dictates how much spectrum and signal strength a transmission can use (see chapter 5).

However in order to evaluate these elements it is first necessary to identify the backgrounds and to evaluate what the main technical considerations are that need to be satisfied in a

PMSE system. As in any system there is likely going to be a trade-off between various parameters that define a wireless PMSE-system.

A typical PMSE setup consists of production, distribution and archive elements. Production is where the performance is made. For audio it consists of microphones, speakers, in ear monitors, mixing tables and one-way and two-way communication systems. The production part isn't limited to just the artists. Audio technicians, but also stage hands may be part of the performance and have In-Ear monitors to monitor the show. An archive stores the performance for later use. Distribution takes the signal and broadcasts it onwards, depending on the situation, this may be from archive or directly from the broadcast. This can be via a broadcast, but also through physical and on-demand media.
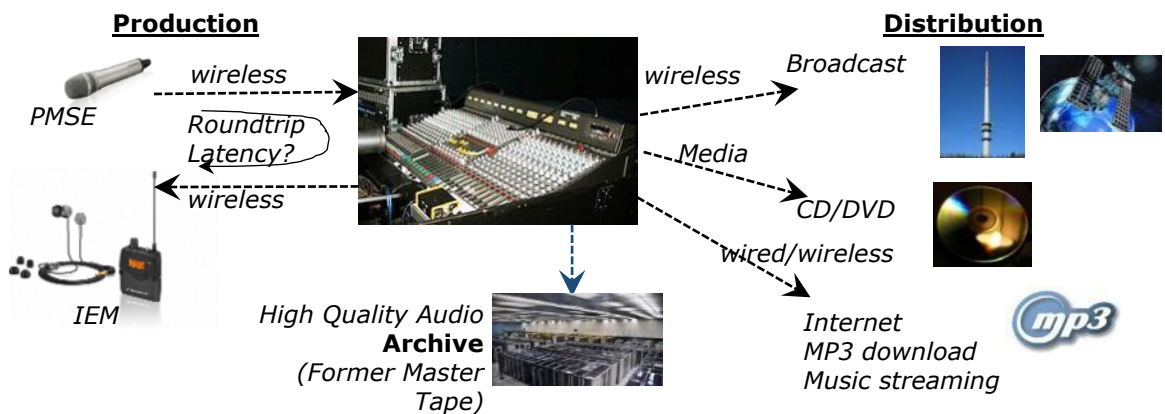

Figure 6 : Typical PMSE Setup (Production, Archive and Distribution)

Wired equipment has the best characteristics in terms of audio quality, power supply, lack of interference etc. Wired microphones and IEMs are therefore common in studio recording. In many environments production costs would be too high if wired microphones and IEMs were used. Wired microphones are often less flexible, require manual labour to hide wires and make production environments safe. Such microphones are in many dynamic situations just unfeasible to use. For this reason there is an ever increasing interest to migrate from wired to wireless. Indeed if studio quality could be supported wirelessly, this would also migrate to wireless. The question therefore is what quality can be produced wirelessly?

A general trend can be observed in communications to "make wireless act wireline". This is also the scope of "convergence" to less distinguish whether bits are transported wireless or wired. For example the Internet protocol works the same over wired and wireless networks and even in lower layers, WiFi borrowed several concepts of Ethernet. In the end it should just be a question of service and not of transport (wired/wireless). PMSE is unique and challenging in its requirements. These requirements are discussed in the following along a typical PMSE set-up as depicted (Figure 6)

An artist is holding a microphone and simultaneously carrying an IEM. The voice by the artist is mixed with e.g. instrument sound and played back to the artist's IEM. By that, the artist can align his pitch and rhythm in singing to the pitch and rhythm of the instruments. It will ensure that the singing has no frequency deviation (pitch) and no rhythm deviation from the instruments. An instrument can always be perfectly tuned to a standard pitch of e.g.

a'=440 Hz, but a singer has to align his pitch to the instrument sound. There are barely artists that have an absolute pitch (absoluut gehoor). So it is of utmost importance that an artist hears his voice together with the instruments. That mandates that the IEM be connected to the mixing console. Often the singer receives a specific mix with the sound from the drums and lead instrument (e.g. piano or flute) pronounced to keep in rhythm and to keep in pitch. Just feeding back the singer's voice to her IEM is not sufficient. The IEM has to deliver a specific sound mix to the artist.

3.2 High availability

For a front singer of a performance, it is required that PMSE works highly reliable, meaning that drop-outs are not tolerated. Furthermore with broadcasting and electronic news gathering when capturing historic events, they cannot be repeated, nor can a politician be requested to repeat his/her statement once again due to problems in recording/production. The unquestionable availability of PMSE service is an absolute must.

On the contrast in cellular networks and other systems "error concealment" is done, where the listener receives some artificial signal to bridge a drop-out. Also retransmissions can be requested. In some applications of PMSE drop-outs and latency are accepted, but only to a few ms. On the contrast in cellular some latency is common and during a cell handover e.g. in GSM two frames of 20ms, so 40 ms drop-outs are very common.

Furthermore it has to be considered, that there are multiple distribution paths. During an event sound is sent to an archive (earlier called Master tape). From this archive multiple formats are derived, e.g. a high end Audio DVD or a highly compressed mp3 download. *What's not captured in the archive during production, cannot be delivered on distribution later.* This means that if there is a drop-out on production, it will be present in any distribution.

Considering e.g. Eurovision Song Contest. The performance of a singer is distributed over around 250 Mil spectators. Drop outs on distribution may be acceptable as only part of the spectators is affected, but these should not be present on the archive. Significant business is also present on further distribution channels, not only the TV life stream. Songs and interviews from artist can be downloaded. CDs/DVDs are being sold.

The archive typically stays for long. The Beatles e.g. mixed mono sound at their times, however when Stereo came up the archive tape was revisited and LPs with "Stereo enhanced" came up the market. In the last years also a Dolby 5.1 mix was derived from the master tape. What is meant by this is that the recoding in the archive is typically done in a way that later new formats and mixes can be derived.

3.3 Low latency

The latency from the PMSE microphone to the PMSE IEM is a critical parameter. When carefully looking at artists without IEM one can see that singers like to stand close to the lead instrument, e.g. first string or piano. Ideally all sounds would arrive at the same time, so that the sound of the voice of the singer as conducted through bone, arrives directly with

the sounds of the instruments. However in practice latencies of 4ms are considered workable. Assuming speed of acoustics at 330 m/s 3 to 5 ms equals about 1 to 1.7 m. However playing with In Ear Monitors creates additional difficulties. Artists hearing their own voice back over the IEMs will need that voice to be in sync with the vibrations conducted via the bones to the ear. Most literature suggests that artists prefer max 3-5 ms roundtrip[11] [3] latency for their IEM. Over 10 ms is generally deemed to be unusable. This puts a significant demand on wireless IEMs and is often cited why digital wireless versions of IEMs are not preferred. The latency added to the total chain by digital encoding of the IEM path would add too much, so that the total latency goes over the critical threshold of 5ms. In Ear Monitors are generally wireless. This allows the artist to move freely on stage. So digitisation is less adopted for IEM.

For comparison some other latency figures are given. In telephony users find a 200 ms max latency acceptable, accounting e.g. for 120 ms in GSM cellular plus 80 ms on fixed line. The GSM cellular system has an physical layer for transmission, where a 20 ms frame is spread over 40 ms implying already 2x40=80ms delay one way for transmit and receive solely from the physical layer/channel coding, let aside source coding. Larger latencies are visible sometimes in news programs on television. Having a foreign country reporter via satellite link means 500 ms total delay for satellite uplink plus downlink as the satellite typically is at geostationary orbit at 36000 km height above earth ground. The interaction between the presenter and the journalist in the field becomes difficult and the audience finds the interview hard to follow. The effect of latency on microphones and in-ears is similar in its effect on perception. If it isn't in sync, both the musicians and the audience will notice something is not right.

Low energy consumption

PMSE devices at the artist like wireless microphone, IEM, body pack transmitter for instruments and so on operate from battery. Often artists are dressed in a specific way and carry tight-fitting costumes and intensive make-up. The battery therefore has to be activated and fitted often before costumes and make-up are put on, which particularly in musical theatre can be hours before the actual show is. Similarly many musicians sound-check hours before they go on stage and they will often leave their equipment active until the show, allowing technical staff to continue to monitor spectrum, but also not to "loose" their radio slot. Therefore battery capacity has to be sufficient, but large form factor batteries, mics and radio hardware are not accepted from artistic performance view.

PMSE equipment at artist side is expected to be "invisible". Technically this implies that the transmission scheme is energy efficient and the electronics are small. From circuit design point of view analogue processing typically is more energy efficient than digital, but being energy efficient and small may limit the radio characteristics. The conversion of an analogue

---

[11] See for example Report ITU-R BS.2161 (11/2009) Low delay audio coding for broadcasting applications, which mentions a less than 3ms requirement based on tests whether experts find the audio quality acceptable. Manufacturers mention similar latency requirements ie http://soundhub.audio/shure-whiteboard-digital-wireless-latency-explained/. Some academic studies, such as The Effects of Latency on Live Sound Monitoring by Lester and Boley suggest some performers might be able to work with higher latencies, however as for example a saxophonist and singer do require low latency, their requirements determine the demands of the system.

microphone capsule signal to digital and its digital processing implies a significant power consumption, that is why a strict analogue wireless microphone (analogue PMSE) is more energy efficient than a digital one. However advances in battery technology and digital communication, in addition to potential low power modes in digital microphones have lead some in the industry particularly from manufacturers to claim that there is no distinct advantage anymore or even say that digital has an advantage.[12] However, more practically inclined interviewees do still mention energy consumption as an important advantage of analogue wireless microphones. Even if digital transmission allows for less RF power, this does not mean a smaller sized RF power amplifier as a power amplifier has to be sized for peak power.

An example: If digital transmission allows for 3 dB less power (cutting RF power to half) this means 3 dB less average power. But digitisation typically comes along with a crest factor contained digital modulation of e.g. 7 dB (factor 5) Crest. In the end the RF power amplifier has to be sized +7-3=4 dB larger, as analogue PMSE uses constant envelope FM. Also digital modulation formats require higher linearity from the power amplifier than constant envelope FM with analogue PMSE. Where a Class-C is sufficient for analogue FM, digital modulation may need Class-A or Class-AB. Class-C operation of RF power amplifier is more energy efficient than Class-A/AB.

3.4 High Quality

It can be observed that on the consumer and thus distribution side there is an increased demand for quality. Artists to some extent are however the most important in how well a system is perceived to perform and they do require high quality audio, often refusing to perform without their favourite microphones and IEMs. Consumers want immersiveness implying high quality video and high fidelity audio. Immersiveness is tied to Video and Audio Quality. Once again, the statement holds: *What's not produced, cannot be distributed*. Quality is directly related to how well the audio is transformed to an electrical signal and back. This starts and ends with how well a microphone captures a sound or how well an IEM can reproduce a sound. However in between there is the mapping of the audio signal either on an analogue or a digital transmission scheme. Current wireless systems do not achieve the same level of quality as high end studio wired systems because the wireless part is limited to a spectrum mask and latency, that limits the sampling rate and dynamic range. To some extent it is doubtful whether they need to as their primary audience is listening to it live and can't hear all the subtleties. In addition sound engineers have to tune the audio mostly for the live performance and not for storing it for later mixing. But what isn't there can't be used. If all audio channels are recorded to an archive a different mix can be derived later.

---

[12] Shure states that digital is more energy efficient. http://blog.shure.com/an-overview-of-digital-wireless-microphone-systems/

3.5 Conclusion

The requirements for PMSE-audio are such that they have to support both the artist while doing the performance and allow for recording and/or transmission of the recorded music. High availability with no drop outs, low latency, low energy consumption and high quality, while still compliant with regulations on what spectrum mask to use are therefore all requirements that are important at the same time. A trade-off is hard to make. A well-known "joke" design guideline in engineering is that it is only possible to get two of the three parameters; good, fast or cheap, in the same system [4].[13]

In the following chapters these trade-offs will be evaluated. Will digitisation allow the same or better quality to be used? Will digital encoding allow low latency communication? Can digital transmission and encoding work with a low energy constraint. In chapter 4 the report will evaluate what the information theoretical aspects of audio encoding for PMSE are. Chapter 5 will evaluate how it is transmitted given the constraints mentioned in this chapter.

---

[13] See for example IETF RFC1902, where it is mentioned as a design concern for all IETF-standards.

4 Information theoretical aspects

The previous section showed that the requirements for PMSE audio are high availability, low latency, low energy and high quality. This section focuses on how the audio is captured and transformed. If there are ways to significantly reduce the representation of the audio through digital source encoding, compared to analogue representations, then this might lead to significant benefits if such benefits can be sustained in transmission. This chapter does this by reviewing how sound is encoded and what parameters (particularly latency) are important

In order to answer the question on whether digitisation can lead to significant benefits in compressing and transmitting audio the report looks at communication model and how this is different from other communications systems. It then evaluates how audio can be encoded given the requirements specified in the previous chapter. The encoded audio also needs to be transmitted. This puts a special set of restraints on the way the audio can be encoded. And in PMSE the latency constraints add a further level of complexity to the information theoretical aspects.

The conclusion of this chapter is that digital source encoding cannot deliver a significant reduction over analogue in PMSE. The latencies involved greatly limit the benefits digital can have compared to situations which are less time sensitive.

4.1 Communication model

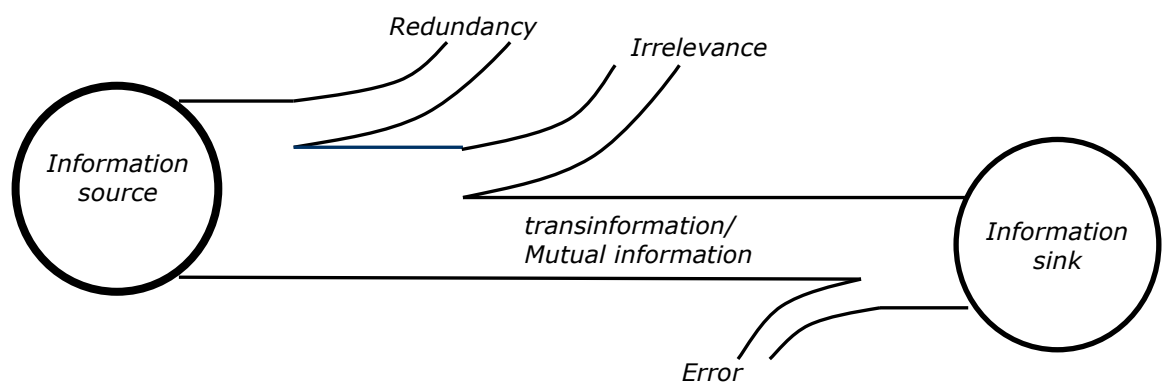From an information theoretical view PMSE is very unique (Figure 7)

Figure 7 : Information theoretical model

In most communication models there are senders and receivers and they are a distance apart. When Marconi made his experiments on wireless telegraphy his aim was to bridge the maximum distance (1901 first wireless transatlantic connection). PMSE is different in this aspect as the artist wants to be able to hear himself and the band. The information source (mic at artist) and the information sink (IEM) are collocated. This is not the case with other systems. Indeed in telephony quite a lot of effort is put into making sure the speaker doesn't hear himself back, other than through bone conductivity. The latency makes speaking quite hard if a person hears him-/herself back through another source. In PMSE it is an absolute requirement that the artist can hear him-/herself back over the speakers or IEMs together

with the instruments. Otherwise it is impossible to keep pitch and rhythm. This results in high demands on low latency performance.

At the information source redundancy can be stripped off without loss of information. Any compression scheme for any sensor at first eliminates redundancy. What can be debated is the irrelevance. Of course, the more irrelevance is identified and stripped off, the less transformation. Minimizing mutual information is attractive as it defines the capacity of the link needed and thus the spectrum amount occupied.

The question however to be asked is, what is irrelevant? Does the transmitter of information know what is irrelevant for the receiver? For example when making or sending a picture it matters whether it will be shown on a smartphone screen as a small icon or whether it is printed to billboard size. In the context of PMSE the question can be reformulated. Do we know what distribution channel is explored - at the time of production? This is typically not known. If we consider above example of the Beatles. Did the Beatles know at the time of recording that later a Stereo enhanced mix or a Dolby 5.1 will be derived? No, they didn't know. Therefore the music has to be captured at the highest quality possible: <u>What's not captured in the archive cannot be distributed later.</u>

This is a fundamental question in cognitive science. What is relevant? When listening to a violinist play, should we be able to exactly how the bow is moved? The average listener may not be able to distinguish that, but for the aficionado it is of the utmost importance. If there is an audience on distribution channel that can hear the fine details of course less irrelevance can be stripped of. If on the other hand there is an audience on distribution channel with an untrained ear and hearing music e.g. in a car or at a bus stop with a lot of disturbing noise they wouldn't recognize if more irrelevance is identified.

So it has to be concluded that the audience on distribution should determine what quality is to be produced. But during production it is not known, who will be on the distribution channel. A trained or an untrained ear? Is the audience listening on a headphone or as background music in a shopping mall? The answer could well be both. And how the content in the archive is used in future, nobody knows. That is why some propose to do no compression at all on production and capture plain audio in the archive. This will allow derivation of any quality level on distribution later.

4.2 Audio encoding and Compression

This paragraph will evaluate the elements of audio encoding and compression in a PMSE environment. How many bits to use and how often to sample. The difficulties with using multiple forms of encoding and how data can be compressed in digital and analogue audio.

4.2.1 Audio encoding

Audio registration requires the audible signal to be represented in a different way for storage and/or transmission. This can be done in an analogue or digital fashion. Analogue aims to represent the analogue waves as precisely as possible. Digital requires a conversion of analogue to digital first. Analogue Audible audio has a range of roughly 110dB. In order to represent this in digital audio an 18 bit code is necessary just to encode the sound. An

additional 6 bits are used as margin to allow for level adjustments. Therefore a 24 bit signal is used to encode audio. This audio has to represent roughly 20 kHz of bandwidth from the lowest tones to the highest. In order to correctly sample this roughly twice the sampling rate is necessary for good quality audio or 44.1 kHz. 18bits at 44100 times per second is 793 kbit/s and at 24 bit this results in 1.1Mbit/s. Studio quality is 24bit at 96kHz which or around 3Mbit/s.[14] This is per channel, so if a sound engineer would want to record all members of an orchestra individually this would have to be multiplied. No matter what type of registration is done, some of the original sound is always lost in registration.

In order to store and transmit encoded data compression is often used. Lossless and lossy compressions are both possible. The first will shrink the data, but in such a way that the original data can be fully recovered. This works because a large quantity of the data is redundant/repeated. Lossy compression removes more of the data, so that the original cannot fully be restored, but if done well in such a way that when the registered data is replayed the losses are not significantly noticeable. Redundancy in the data allows algorithms to reduce the data in size. Time is however very important in determining the maximum compression of an audio recording possible. The shorter the time that is recorded the less redundancy there is in the recorded data. Also shorter times reduce the algorithm in the amount of passes it can make to compress the data. This is particularly a problem in PMSE as the latency constraints are very strong and therefore influence the ability to reduce data through compression.

4.2.2 The problem of cascaded codecs / cascaded compression

On distribution a compression scheme (codec) is applied in most cases. Distribution via Mp3 download, DAB, DAB+, audio inside DVB-T - this all includes a lossy compression scheme. If compression is also applied on production a cascade of two compression schemes is faced. There is a serious risk of artefacts if multiple compression schemes are cascaded. If one codec determines part of a sound as less important and doesn't convert and transmit it, a following codec may throw away another part of the remainder. This cascading may lead to complete loss of the signal and may make the communication unusable. The cellular industry is heavily exploring compression schemes for voice e.g. Codecs EFR Enhanced Fullrate, HR Halfrate, AMR Adaptive Multirate codec. But the cellular industry carefully avoids cascading of multiple codecs by routing compressed data in their network in the case of a mobile to mobile call even with different cells (Figure 8).

---

[14] Some studios record in even higher fidelity, recording 32bit at 192kHz.
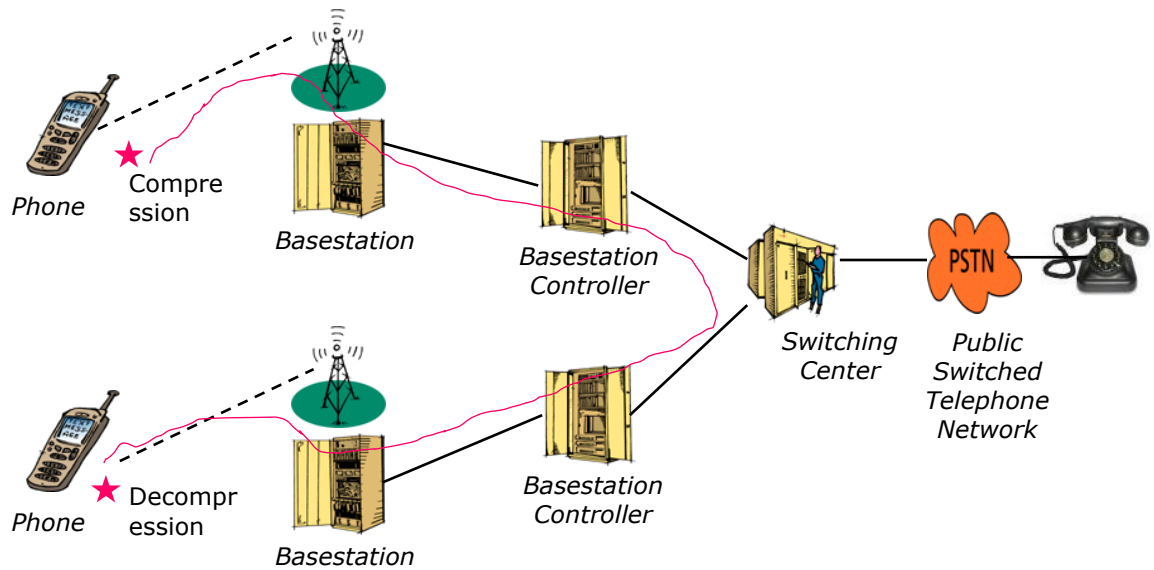
Figure 8 : Routing of compressed data in cellular network -avoiding cascade of two codecs

To get around the problem of artefacts with cascaded codecs only small compression factors are used during production with PMSE on the order of typical 2:1 for live performances, 1:1 in studio environments. For distribution larger compression factors are explored. A quick example. A CD has 500 Mbyte and may contain 10 Tracks. That is 50 Mbyte per audio file. On the internet an mp3 file of 5 MByte may be downloaded which means 10:1. If the file is only 2.5 MByte than the compression factor is 20:1.The original song however may well be 500Mbyte of audio. The unmixed recording of each individual instrument might even be more.

As the codec on distribution is not known during the time of production, no experiments can be conducted whether a certain compression scheme on production would cause artefacts in cascade with the compression scheme on distribution. To be on the safe side some propose plain audio / uncompressed on production.

4.2.3 Compression in analogue and digital audio transmission

Using a standard spectrum mask of 200 kHz as defined by CEPT requires 6 symbols per Hz in transmission. This is equivalent to 64QAM or similar encoding schemes and goes beyond what WiFi and LTE achieve and is a little below the 8 symbols/Hz in DVB-T. Sennheiser for example states that it can reach this quality level, however only at the cost of reducing the range of the system as to achieve this quality level the signal to noise ratio needs to be excellent on the order of 40 dB.[15]

Digital wireless microphones therefore often use a form of compression in order to achieve the needed range. Compression, also named source coding is not something new that comes with digitisation of PMSE. Analogue PMSE also uses a compression scheme named compander system. It is also combined with preemphasis on FM that reflects the fact that

---

[15] Studio quality is 24bit at 96kHz which would demand 30bit to be transmitted wirelessly or around 3Mbit/s

higher frequency tones typically have less amplitude. The analogue compander system is equivalent to a 2:1 compression. An analogue audio dynamic range of 100 dB (near CD equivalent is compressed to 50 dB, so 2:1.

Companders allow a reduction in the amount of spectrum used given the dynamic range. However they have their own artefacts, because there is no way for sender or receiver to adequately know what has happened to the signal when it was compressed and expanded, nor how well the uncompressed and expanded signal relate to each other. Or to put it in terms of a compander system; though it may be known how much the compressor took out, knowing where the compander took what out and adding it back in in the expander is an inexact science, particularly if the wireless medium adds artefacts and errors in transit. A compander system is a nonlinear scheme which is not fully reversible given the bandwidth limitation between compression and expansion on the wireless link (Figure 9).

It is this non-linear nature that some audio engineers in the interviews stated that they don't like about analogue audio in PMSE. An analogue compander can give different effects for different instruments, singers and environments. For example the lower notes are more affected by companding than the higher notes, so the same compander settings may work for a soprano, but not for a bass singer. A digital encoding is more consistent across a range of devices, singers, bands and environments. For a sound engineer it makes it easier to fine-tune that part of the audio chain, so that the performance is consistent.
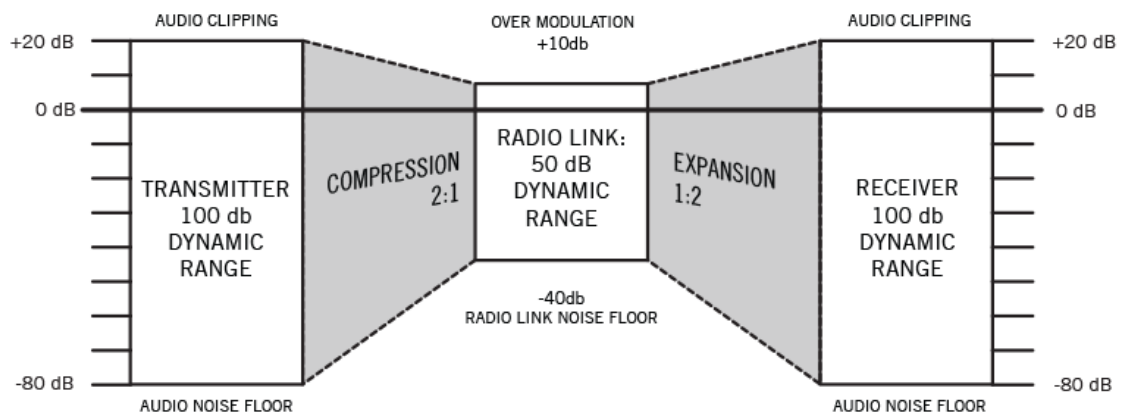

Figure 9 : The working of a compander (source: Shure)

If analogue and digital PMSE both use 2:1 compression analogue and digital transmission within an identical RF channel bandwidth of 200 kHz are par. To some extent digital is even worse on spectral efficiency as there is overhead due to channel training, signalling and synchronisation symbols. Digital therefore fits more strict within the RF channel bandwidth than an equivalent analogue signal. Digital transmission is from an information theory perspective only attractive for higher compression factors beyond 2:1, when the compression starts to overcompensate the additional overhead due to digital transmission. This is typically the case on wireless distribution where higher compression factors are in place. The graph below displays this to some extent (Figure 10).

Figure 10 : Sound quality

The graph is based on the following calculations:

- As a reference point uncompressed CD mono quality is assumed which accounts for 16 bit at 44.1 kSa/s=705.6 kbit/s.
- DRM, DAB and DAB+ figures are taken form literature. They are commonly available. For DRM a compression mode delivering 20 kbit/s is assumed.
- For DAB its data rate of 160 kbit/s is divided by factor 1.4 to account for the fact that a fair comparison on mono has to be conducted. The factor 1.4 was given by Audiolabs Professors as Stereo coding does not simply lead to factor 2.
- For DAB+ its data rate of 80 kbit/s is again divided by factor 1.4.
- Digital PMSE robust assumes a CD mono quality being compressed by factor 4, which leads to 176 kbit/s.
- For the case of PMSE standard a CD mono quality compressed only by factor 2 is assumed, leading to 353 kbit/s.
- For Studio quality I a sampling rate of 96 kSa/s at 24 bit leading to 2.3 Mbit/s is taken.
- For Studio quality II a Sampling rate of 192 kSa/s at 24 bit is taken.

4.3 End to end transmission of compressed data

4.3.1 Overhead in digital communication

In contrast to analogue transmission digital transmission incorporates a significant amount of overhead. Figure 11 shows the architecture of a digital transmitter.
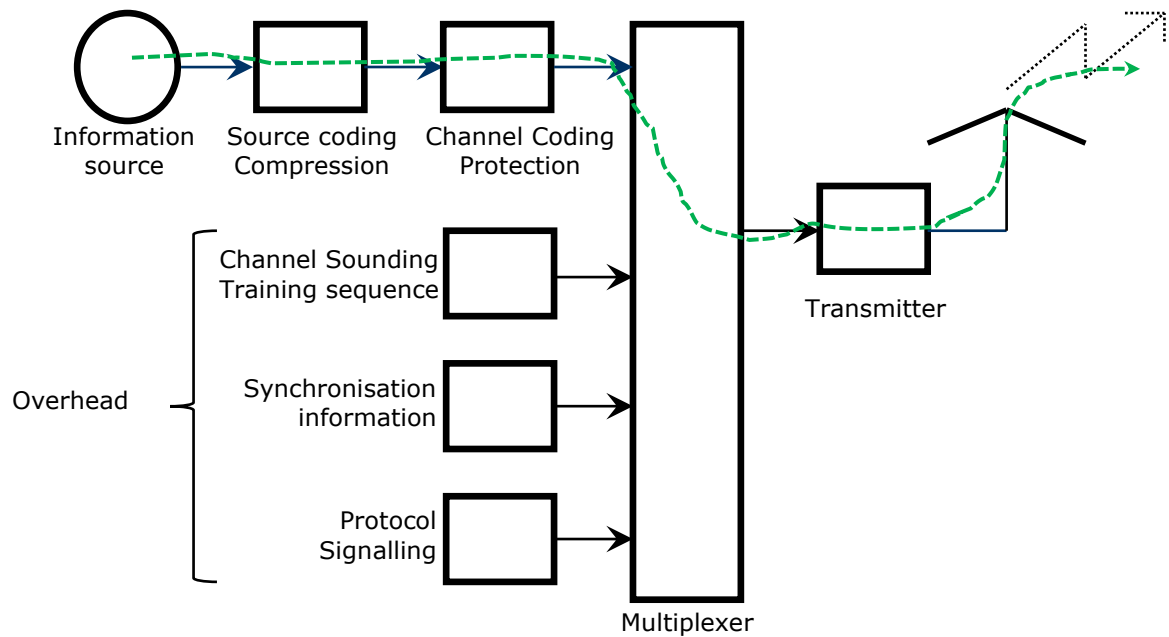
Figure 11 : Architecture of digital transmitter

The data from an information source, e.g. microphone is digitized and undergoes source coding (compression). Then the compressed data is protected by channel coding. But that isn't all. The coded data has to be accompanied by training symbols for channel sounding, synchronisation symbols and signalling protocol. This all plays into overhead, which is not present at all on analogue transmission. Therefore from an information theory point of view on a single channel there is no benefit to a digital transmission if no significant compression can be achieved.
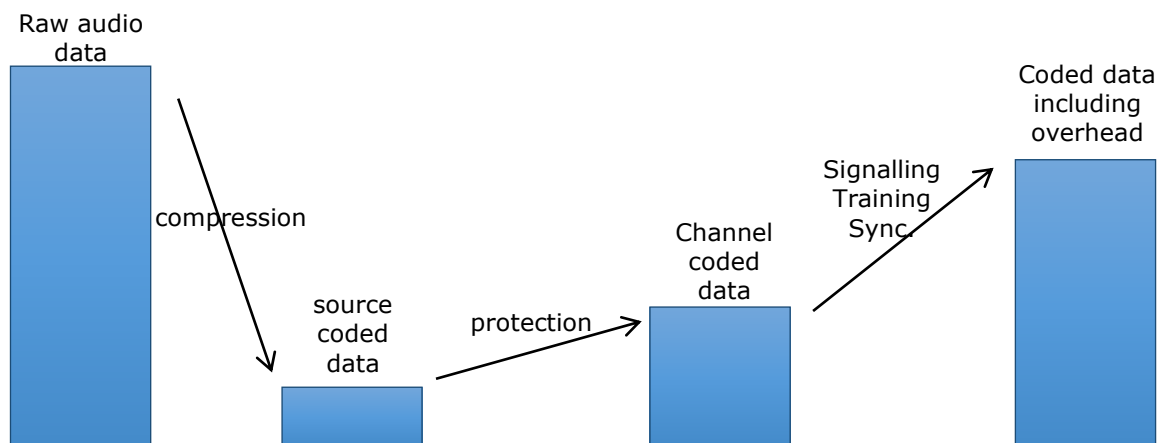
Figure 12 Digital communication link

Figure 12 is highlighting the data amount along a communication link. By source coding we reduce data amount. Then we increase data amount by channel coding. Then we place further overhead on top. From the figure it gets clear that digital pays off only if source coding reduces data sufficiently, which is not the case on low compression factors. In other words it can be said, if compression is not applied at all, there is no reason to go digital to

save on spectrum. Avionic radio for instance where correct understanding is an absolute must for reason of security does not use compression at all and is still using analogue AM modulation.

A good example of the amount of overhead that can be added is VOIP over cellular. The raw audio data is only 3%, the application, TCP/IP and underlying layer 2 transmission are the remaining 97% of the data stream. The conclusion should be that overhead with digital transmission cannot be neglected [5].

To conduct a fair comparison between analogue and digital this has to be based on identical audio quality and identical latency. The robustness improvement by 10 dB for digital mentioned in OFCOM report comes from source encoding as more channel coding can be put on top for protection. Therefore performance comparisons heavily depend on what is considered equal audio quality. Furthermore analogue transmission has negligible latency.

4.3.2 Compression limitations: Shannon bound

The Shannon bound found by Claude Shannon of Bell Labs in 1948 reflects a fundamental limit in communications:

$$C = B \cdot ld\left(1 + \frac{S}{N}\right)$$

Capacity [bit/s] equals Bandwidth [Hz] times logarithm dualis of 1 plus SNR. There is no transmission scheme beyond this bound. From this law it makes no sense to name a maximum data rate "bandwidth". It must be named capacity.

The Shannon bound is related to the data after source coding, not to the data before the source coding. This is often misinterpreted. The Shannon bound does not make any statements on source coding. It just describes the bound for transmission including channel coding.

The Shannon bound is an asymptotic bound for infinite block length implying infinite latency. For limited block length, thus limited latency the capacity is less or alternatively more SNR is required for identical data rate. Under a constraint of low latency as for PMSE for a wanted capacity more SNR is needed. The low latency constraint can easily account for a 10 to 15 dB hit in SNR beyond the Shannon bound. Therefore it is not allowed to calculate spectrum need for PMSE by using an unadjusted Shannon bound.

Analogue transmission is not per se inefficient compared to digital one. Analogue SSB (Single Side Band Modulation) is 0 dB close to the Shannon bound. Such an analogue scheme could never be outperformed by any digital transmission scheme. Digital transmission always suffers from additional overhead required.

The following graph (Figure 13) shows the capacity [bit/s] for a PMSE like scenario under the assumption of an equivalent receiver noise bandwidth of 180 kHz. This "equivalent receiver noise bandwidth" typically is narrower than the channel width of 200 kHz due to the fact that the receiver's filter chain composed out of analogue and digital filters is optimized for highest sensitivity. A filter also neither is rectangular in frequency domain as this would imply infinitely long impulse response. The equivalent noise bandwidth is a virtual measure for a hypothetic rectangular filter that would pass the same total noise power as the real filter

chain. If a filter is made too narrow, not all of the wanted signal's power is captured, degrading sensitivity. If on the opposite the filter is made too wide, too much noise is captured also degrading the sensitivity. Therefore the receive filter's shape is carefully optimized. For the following calculations an assumption of 180 kHz equivalent noise bandwidth is made.
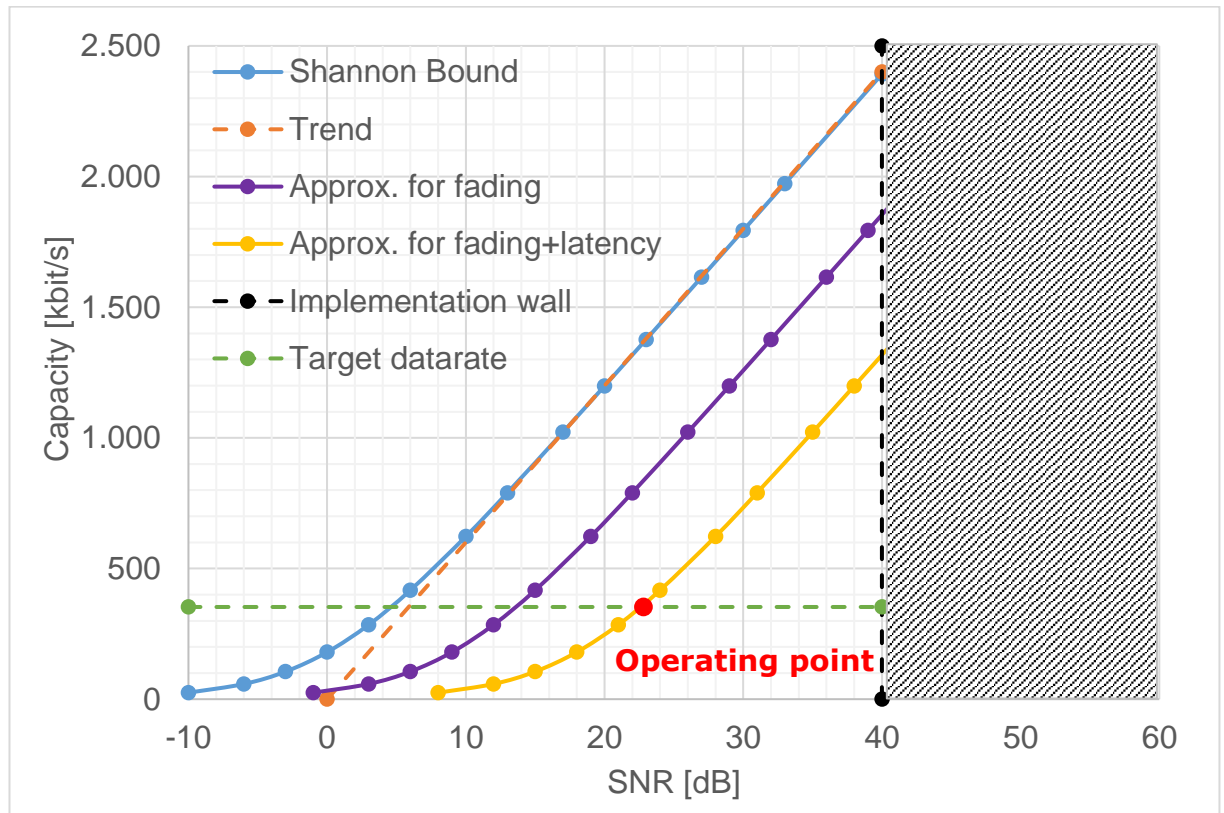

Figure 13 : Shannon bound plus approximations accounting for fading and latency constraint

The fundamental Shannon Bound assumes that there is no fading and that there is no latency constraint, thus block length in transmission would be infinitely long. The Shannon bound is an asymptotic bound when block length goes to infinity. Because of fading conditions in PMSE use and the very strict latency constraint on the order of 1 ms one way, an educated guess was made to shift the fundamental Shannon bound two fold. As a hit due to fading 9 dB was assumed. Furthermore as a hit due to latency constraint again 9 dB was assumed.

Now considering a microphone capsule offering CD like quality Mono with 16 bit resolution at 44.1 kSa/s would deliver 705 kbit/s net. After a 2:1 compression one would have to transmit 353 kbit/s. Looking for the crosspoint now delivers that the operating point of PMSE system is at around 22 dB minimum SNR. Such an operation point is fairly high. For comparison consider GSM, which also uses a 200 kHz channel, can work at 9…13 dB SNR.

This much higher SNR operating point compared to other wireless systems explains why PMSE is so vulnerable by interference. It explains why intermodulation products have to be carefully managed in PMSE frequency allocation plans.

Even if a 4:1 compression is selected, the data rate to be transmitted would reduce to 176 kbit/s, but still the SNR operating point would be very high at around 18 dB.

The Shannon bound furthermore also assumes that there is strict Additive white Gaussian Noise (AWGN). However in reality a receiver faces a superposition of noise and interference, whereby the interference typically dominates. Now it is well known in receiver design, that a receiver is more robust against pure random noise rather than structured signals like interference, which easily accounts for another 1…2 dB hit. Also due to the fact that the receive filter is not flat, the noise gets spectrally shaped, it is no longer white, which also accounts for a hit by 0.5 dB. This all together explains why the operating point of PMSE so much deviates from the simple Shannon bound and why digitization cannot get PMSE to the Shannon bound. There isn't so much room for improvement by digitization as the simple Shannon bound predicts.

Fairly speaking the graph presented here reflects an approximation. For detailed studies, careful simulations would have to be conducted making certain assumptions about fading channel properties and channel coding schemes. There have been advancements especially for the case of latency constraints to apply channel codes that perform better under low structural delay and perform worse under large block length, like e.g. LDPC codes (Low density parity codes). In literature approximations for the theoretical bounds under varying latency constraint have been derived and discussed[16]. Research in this area is still ongoing to support also the definition of 5G URLLC services (ultra reliable low latency communication).

The graph further contains an "implementation wall". In practical designs the maximum SNR is limited to about 40 dB due to phase noise and non-linearities existing in every receiver. This limits the maximum data rate to about 1.6 Mbit/s in a 200 kHz channel.

4.3.3 Robustness: Channel coding, interleaving and block length

Channel coding often includes interleaving, which means that neighboured bits after source coding are spread widely in time in a pseudo random like fashion. It is an attractive technique against bundle errors, that make channel decoding fail. Interleaving can be treated as some sort of temporal diversity. However if there is a stringent latency constraint as in PMSE, the block length also is short on a ms Scale. If block length is short compared to channel coherence time, which is on the order of 40 or more ms, then no gain can be drawn

---

[16] Christoph Rachinger; Johannes B. Huber; Ralf R. Müller, Comparison of Convolutional and Block Codes for Low Structural Delay, IEEE Transactions on Communications, Volume: 63, Issue: 12, Year: 2015, pp 4629-4638

Christoph Rachinger; Ralf Müller; Johannes B. Huber, Low latency-constrained high rate coding: LDPC codes vs. convolutional codes, 8th International Symposium on Turbo Codes and Iterative Information Processing (ISTC), 2014, pp 218-222

Thorsten Hehn; Johannes B. Huber, LDPC codes and convolutional codes with equal structural delay: a comparison, IEEE Transactions on Communications, Volume: 57, Issue: 6, 2009, pp 1683-1692

from interleaving. In cellular Transmission Time Intervals (TTI) of about 2 ms are used with no interleaving.

Interleaving makes digital data transmission more robust, but increases latency especially as the interleaver's temporal length hits the latency budget twice, at TX and RX side.

4.3.4 Separation theorem

Figure 14 shows a complete communication link composed out of source coding and channel coding on transmit side together with their counterparts on receive side.

The Block diagram shows a clear split between source coding (compression) and channel coding (protection). In digital transmission normally the so-called "separation theorem" is assumed, which means that source coding and channel coding can be optimised independently of each other. Strictly speaking this separation is only allowed for infinitely long blocks thus infinite latency. However, considering the requirement of ultra-low latency, the separation theorem no longer holds. Already in GSM transmission where the latency is just 120 ms one way, the separation theorem was not assumed. The voice coder in GSM produces class-1 and class-2 bits of different importance and passes them to the channel coder individually. The Class-1 bits as they are more important get a heavier code protection than class-2 bits. Considering that already in GSM with 120 ms latency one way the separation theorem is not explored, it is obvious that in PMSE with a much more severe latency requirement of 2 ms one way, the separation theorem cannot be assumed. This would mean that in PMSE surely source and channel coding should be optimised together.

Regarding feedback to the artist by IEM this immediately implies that channel and source decoding has to be done in front of mixing console's input to completely reconstruct the microphone audio signal as plain audio before it can get mixed with instrument sound. Mixing is only possible on plain (uncompressed) audio, not on compressed audio.
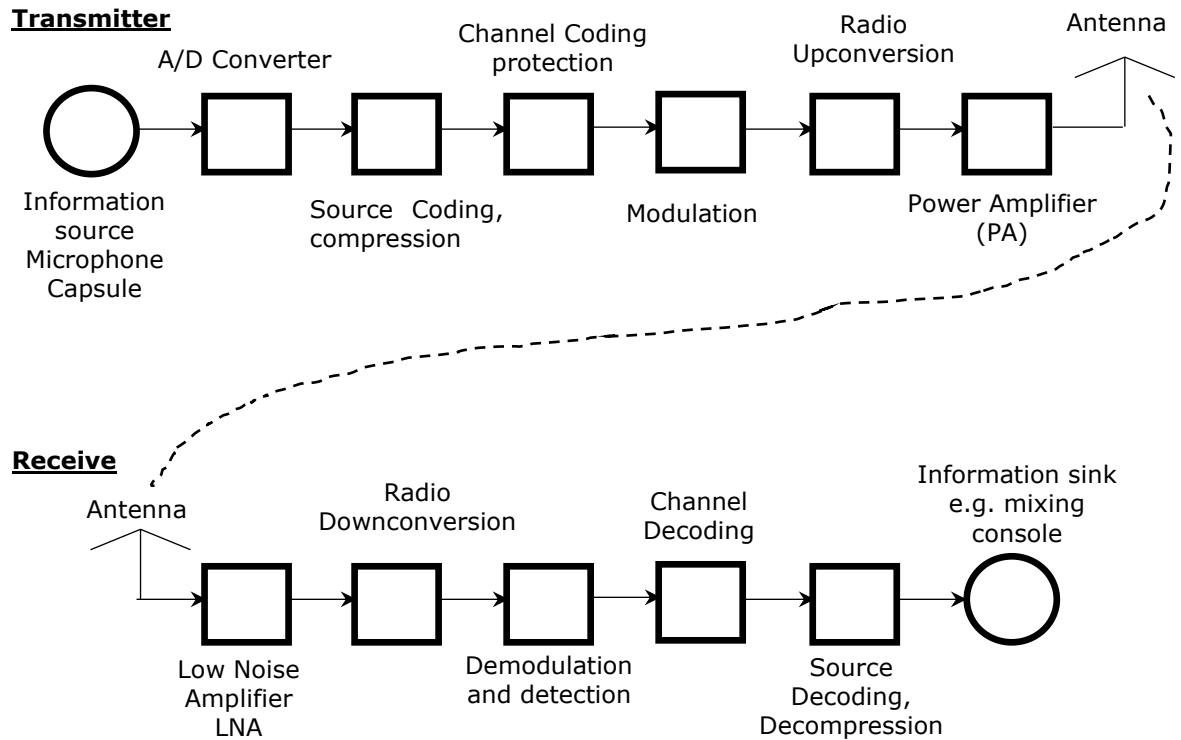
Figure 14 : Communication link

Then after mix, inserting the instrument sound, a similar scheme on the PMSE IEM link can be applied. Now the latency on mic link to mixing console and from mixing console to IEM add up. Also the risk by cascaded codecs applies. Once again it becomes obvious that usage of plain audio on wireless link to and from mixing console would be advantageous.

4.4 Impact of latency constraints

As stated earlier the PMSE-situation is different, because it wants source and sink to be closely together. Indeed for singers one could state that it wants the electronic sound to arrive at the ear, at the same time as it is conducted through the bones to the ear. That of course isn't the precise requirement, but the demands are large and anything above 10ms is considered very noticeable. 5ms is considered to be the maximum allowable for many applications. This paragraph therefore further evaluates the latency constraint in the context of the information theoretical aspects.

4.4.1 Latency Budget

Several factors play into the latency budget of a digital communication link:

- A/D conversion
- Source coder/compression
- Channel coder/protection
- Digital transmitter, wireless link and digital receiver can mostly be neglected
- Channel decoder/reconstructing send data

- Source decoder/decompression
- D/A converter

Of course in the case of feedback to an artist the latency budget is more or less doubled. Latency of A/D and D/A conversion typically is small and can be neglected. Mixing typically incorporates further delay due to resynchronisation schemes of multiple digital connections.

In digital communication interference cancelation receivers also named advanced receivers get more common to increase robustness. However they explore iterative schemes, which means that robustness comes at the cost of latency. The potential of advanced receivers in the context of low latency communication therefore is questionable.

4.4.2 Latency hit by resynchronisation and resampling

The latency budget for roundtrip from artist microphone to artist IEM also has many other contributors aside of the latency of the digital microphone. As the microphones so far only have a unidirectional link from mic to mixing console, they run unsynchronised to the mixing console. That implies that every microphone and instrument signal received in digital format has to be resynchronised to the clock of the mixing console.

High quality Mixing consoles typically internally run sampling rates of 192 kSa/s. Any incoming sampling rate is up converted to this rate. This rate conversion works even on non-integer ratios. Resampling can account for up to 0.5 ms.

A homogeneous digital environment would avoid resynchronisation. However this demands that the ADC in a wireless mic runs synchronous to the sampling clock in the mixing console. This would be enabled if the PMSE link is bidirectional so that clock info is transported in the direction from the mixing desk to the wireless mic. That's not possible today as a digital wireless microphone is unidirectional these days.

Some companies have established a second wireless network e.g. based on Zigbee to command signalling info to PMSE devices like commands of centre frequency and coding. Time protocols similar to NTP (Network Time Protocol) in TCP/IP could be run.

During the course of the German National Research project PMSE-xG the synchronisation challenge is addressed[17]

For wireless digital IEM there is no problem regarding resynchronisation as the D/A converter in the IEM gets its clock over the air from the mixing console. However digital IEM are not used today as the benefit in terms of sound quality from digital do not outweigh the drawbacks of increased latency when going digital.

4.5 Graceful degradation through scalable source coders

Analogue FM transmission has the nice property of graceful degradation. For 1 dB less RF SNR there is also 1 dB less Audio SNR. This works down to the FM threshold, which is inherent to analogue FM. Digital transmission with channel coding in contrast to analogue

---

[17] http://pmse-xg.de/downloads.html

transmission works excellent down to a certain threshold in RF C/N, but then digital transmission totally collapses. This is sometimes called the "digital cliff".

It is interesting to ask whether something similar like in analogue - graceful degradation - could also be realised on digital transmission and whether this, even if it doesn't decrease data amount, increases the usability of digital. One could use so-called scalable source coders, whose output bits are not equally important. Certain classes of bits exist. This e.g. is done with GSM speech coders. There are class I and class II bits, so only two different levels of importance. There are source coders, which produce bits of multiple importance levels. However the overall compression level is low, so the relevance is questionable. Graceful degradation would allow avoiding the digital cliff, sudden and unexpected drop-out of audio signal when reaching the maximum range. However, graceful degradation would come at the cost of increased spectrum demand.

4.6 The future of audio codecs

As stated before there is a clear distinction between speech and audio codecs for singers and instrument. In speech coders there is a certain speech model that is heavily explored. That is why speech coders achieve much higher compression rates e.g. 22:1 in cellular (AMR codec).

Something similar could also be envisioned for singers. However the problem would be that the coder has to adapt to the singer's specific properties. If a singer would hand over its microphone to another singer or move it in front of an instrument during a performance, source coding would collapse. Sound quality would be terrible. Of course source coder could readapt, but that needs time. Whenever there is a change in the audio source's properties, artefacts would arise and the time for readaptation would start again. Thus, such an option is not practicable. Higher compression would come at the cost of less flexibility.

Looking to analogue compander systems, the compander scheme is selected based on the source's properties. But this is considered a serious inflexibility. Typically weaker compander systems are used but this requires high RF SNR and a good frequency plan. Strong companders are only explored in problematic RF environment.

4.7 Implications of an all-digital solution

In a typical PMSE usage scenario (Figure 15), where an artist caries a wireless microphone and an IEM, two PMSE links are involved. The first PMSE links runs from the microphone to the mixing desk and the second PMSE links from the mixing desk to the IEM.

Now let's assume that one wants an all-digital solution (marked as red cloud). Furthermore one wants to reduce spectrum demand implied by both links through heavy source coding as otherwise range would be compromised. What would be the consequence?

The first consequence would be that source decoding of microphone signal has to be conducted in front of digital mixing desk as mixing is only possible on plain audio. Otherwise the instrument sound cannot be added. It is essential for a vocal artist that instrument sound is mixed with his voice and fed back to his ears, as otherwise the vocal artist cannot adapt his beat and pitch to the instruments.

The vocal artist now faces a cascade of two codecs on the link to and on the link from the digital mixing desk. Even in high performance digital audio systems this will lead to some form of artefacts as the compressed coded signal is of lower quality than the original, the decoded signal is therefore never a perfect representation of the original. This then has to be mixed with other signals and then recoded and sent onwards. Some biases of the original encoding/decoding may combine with the biases to send it to the IEMs to create artefacts. Artefacts implied by the cascade of two codecs will be very distracting for the vocal artist. What comes on top is the delay by the two codecs and their digital wireless transmission. The combination of artefacts and delay is the reason why most artists prefer an analogue IEM. A digital output of mixing desk is converted to analogue and sent via an analogue PMSE link to the IEM.

In addition using analogue IEMs helps solve the round trip delay problem imposed by using an all-digital solution. If encoding and decoding audio adds 2-3 ms delay even in the most advanced professional systems, then the round trip would be a minimum of 4-6 ms, but likely a bit more, in an all-digital solution. This would imply that it becomes 'unusable' from the perspective of at least the singer and even other band members would consider it difficult to use. Analogue IEMs give sound technicians an easier envelope to work with. It might be possible to have an all-digital solution, but the constraints are less when analogue IEMs are used.

The latency budget is dominated by source coding. In order to make the whole digital scenario acceptable from roundtrip latency point of view, source coding would have to be taken out, going for audio uncompressed. But that would need more (double) spectrum.
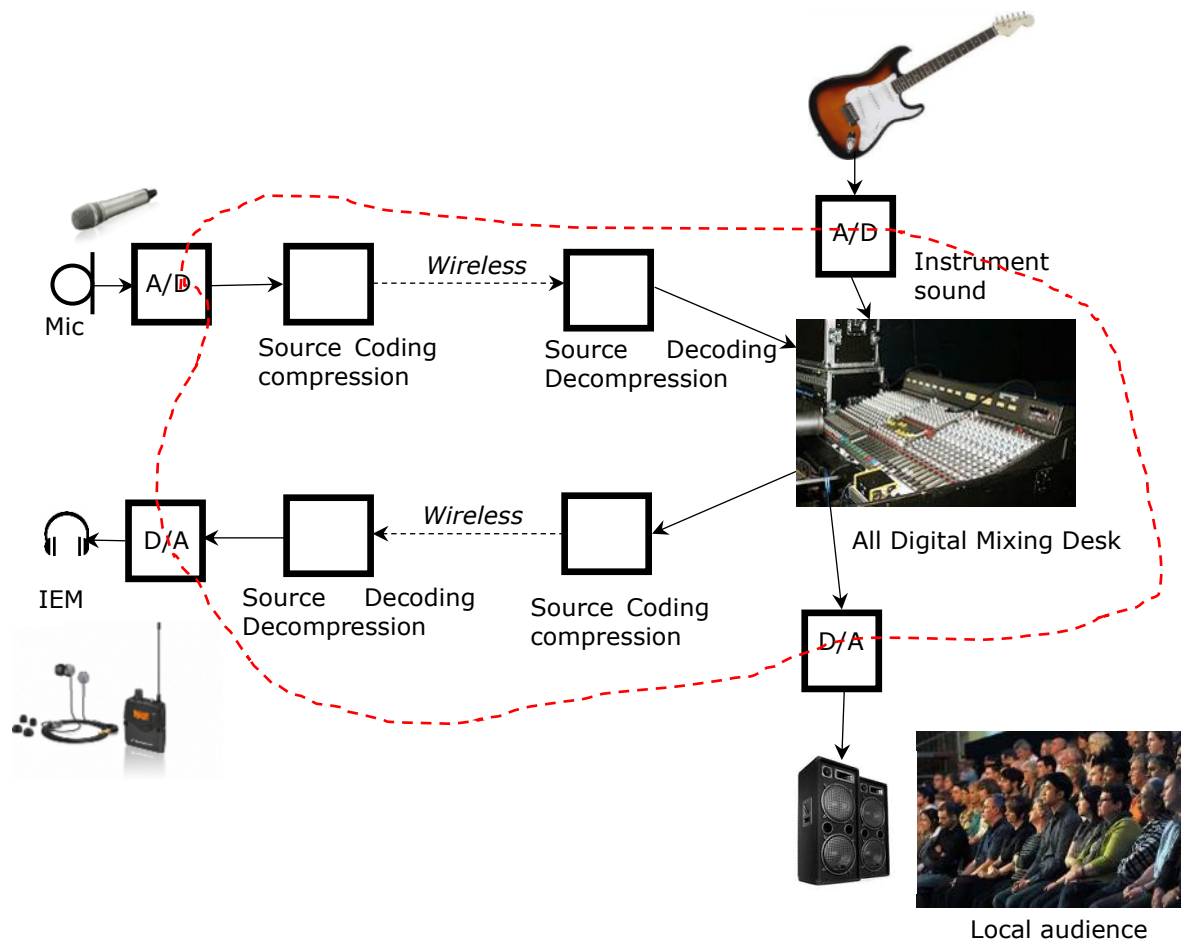
Figure 15 : Feedback to an artist

4.8 Conclusion on information theoretical aspects

Information theory doesn't give a basis for expecting digital signals to use less spectrum than analogue signals in a PMSE environment. Both systems are roughly on par with a 2:1 compression ratio if distance and the use of companders is factored in. It is possible to achieve a higher quality using digital at the expense of distance and with high requirements on SNR. Particularly the low latency requirement results in an inability to reduce the data that needs to be sent. There is no time for interleaving or other forms of compression based on speech models that can be explored in other applications with less stringent requirements.

What most industry experts agree on is that the need to use a compander makes analogue less desirable to digital. It's non-linear and non-consistent performance make it harder for sound engineers to work with analogue systems. They therefore accept the drawback of digital with regards to the digital cliff and latency and appreciate the consistency in sound quality. The additional benefit of being able to do high quality audio is appreciated for some applications, such as lead singers. As a result it is likely that digital audio will increasingly be used.

5 Radio aspects

Analogue and digital transmission systems have different spectral characteristics. Both represent a conversion of some signal onto a bearer. However their effect on spectral efficiency and on interaction with other signals are quite different and therefore result in the more significant differences between the two techniques, much more so than the differences between analogue and digital audio encoding. It is here where the more fundamental differences between analogue and digital wireless microphones and IEMs lie.

As the previous chapter the question starts with the fundamental properties of radio transmission of audio. What is efficient use of spectrum? How is that evaluated? The chapter then evaluates how the critical parameter of reliability and robustness can be achieved in a PMSE environment. The report then takes the spectrum mask that is currently specified for PMSE and evaluates how this affects the density of the number of devices that can be used in parallel. This theoretical number is however greatly limited because PMSE-transmitters are spaced closely together and pick up and retransmit each other's signal, this is known as intermodulation. It is this intermodulation problem that is the basis for predictions that the there isn't enough spectrum available for the PMSE-sector. We will show that fundamentally there are no differences in intermodulation between digital and analogue transmission. Indeed it is the hardware used that determines whether intermodulation is a problem.

5.1 Spectral efficiency

The term "spectral efficiency" is used in very different ways. It is therefore important to define how spectral efficiency is used here. From information theory spectral efficiency is related to the data after source coding, meaning that source coding/compression cannot improve spectral efficiency in a strict sense. It could only do this if less spectrum is needed to transmit the same data. In this paragraph the various aspects of spectral efficiency are discussed and how they relate to PMSE are explored, particularly as the goal is to identify how many PMSE-devices can be held in the bands designated to them.

5.1.1 Spectral efficiency of what?

The term 'spectral efficiency' is used for a number of different cases. It is very important to define and distinguish these cases, as 'spectral efficiency' in one case does not necessarily mean 'spectral efficiency' in another case.

The following definitions are taken from publications by Prof. Friedrich Jondral from KIT Karlsruhe, who is known as an expert in SDR:

a) Connection spectral efficiency: Spectral efficiency of a point to point connection

This is understood as the number of bits transported within a second and within a given bandwidth across a Point to Point link. It is given in [bit/s/Hz]. It can be increased by higher order modulation and MIMO techniques. The usage of higher order modulation for higher data rate requires more SNR, whereas the MIMO way just requires propagation channels with multiple eigenmodes and multiple antennas at both sides of a link.

b) System spectral efficiency: Spectral efficiency of a wireless communication system

This is understood as the number of bits transported within a second and within a given bandwidth summed over all users normalized to area. It is given in [bit/s/Hz/km^2] aggregated over all users. It can be increased by tighter packing of signals, smaller cells and more dense placing of communication nodes.

c) Overall spectral efficiency: Efficiency of spectrum use,

This is understood as the number of bits totally transported within a second and within a given bandwidth summed over all users and systems normalized to area. It is given in [bit/s/Hz/km^2] aggregated over all users and all systems. It can be increased by implementation of co-primary and secondary systems, which make use of resources actually not in use by primary systems. So it reflects an opportunistic access to temporarily and locally unused spectrum.

5.1.2 Increase of spectrum efficiency or information efficiency

An often used example is the increase in spectrum efficiency for digital television. When one analogue TV channel was substituted by DVB-T, one RF channel that originally carried one analogue TV program now could carry 4 to 6 digital TV programs. This was only possible due to source coding (compression) of digital video signals and was not the benefit of digital transmission itself. Indeed overhead had to be accepted for digital transmission like synchronisation symbols in OFDM. The replace of one analogue TV program by 4 digital ones was named a spectral efficiency increase due to digital transmission but strictly speaking it is solely the benefit of source coding. So the channel still was 8 MHz wide and the same number of channels were in use. The data carried is now digitally compressed and so it became possible to view more television channels over the same number of radio channels. Therefore strictly speaking the informational efficiency had increased while the spectral efficiency had remained the same.

With digitisation of PMSE the case is such that the same signal is transmitted in the same spectrum mask, however the analogue signal is transmitted digitally. In PMSE there is already a starting point incorporating analogue source coding by a compander, which means that with a transition from analogue to digital, the benefits of digitisation are low. It is not that more information can be handled over the same spectrum, but that the same information is handled differently. In one spectrum mask one microphone or IEM-signal can be held. It's not as with DTV that the one channel can carry 4 microphone signals. However there could be possibilities to use less spectrum overall to carry 100 signals in PMSE.

5.1.3 Spectral efficiency for PMSE

In the context of wireless PMSE manufacturers and users try to optimise each different definition of spectral efficiency. Increasing the use of compression codecs should not be considered spectral efficiency (A) as this is only decreasing the number of bits to be sent and any connection by increasing the number of bits/Hz to achieve higher quality for the audio signal within the same channel is used as we've seen. The use of MIMO so far hasn't been mentioned, by manufacturers for UHF, but might well be something that is researched. This

would allow for multiple inputs and outputs on the same channel, which combined allow for more bits to be sent over the same channel. However MIMO has a certain inherent fickleness, because it needs multiple propagation paths. Therefore it works better if the user is in environments with multiple paths.

The spectral efficiency of a communications system (B) is increased in modern PMSE systems by allowing a denser packing of signals. Shure has advertised that it can hold up to 63 devices in an 8 MHz channel. It does this by spacing the devices very tightly, reducing the power to 1mW and not operating in a high quality mode. However the location needs to have very little noise in order to function. This mode is therefore limited to conference type situations that are indoor.

In addition to packing more devices in a channel it is also possible to increase the overall spectrum efficiency (C), the *efficiency of spectrum use.* What the system then does is to use spectrum that is in use by other applications. For example some manufacturers explicitly offer the possibility to use spectrum that is also in use by DVB-T at the same time. This isn't recommended for amateurs, nor in outdoor situation, but in a well shielded environment such systems may allow the use of frequencies otherwise considered unusable.

To increase the "efficiency of spectrum use" some research is done to further increase this by working in UHF spectrum that is now used for cellular. PMSE typically reflects a temporary and local usage. Coordination with other systems therefore is a promising approach to improve "efficiency of spectrum use". And indeed this was done in past times when PMSE was coordinated with TV use. Now having large parts of UHF spectrum dedicated to cellular, coordination of PMSE with cellular no longer is possible. However other ways of coordination can be envisioned like integrating PMSE into cellular, (see paragraphs 4.4 and 5.5 for more information about the PMSE xG project).

5.2 Reliability and perceived robustness of transmission quality

In every interview the reliability problem of digital transmissions was discussed. Digital systems suffer from what is known as the "digital cliff" (Figure 16). It works, until it doesn't work and then when the signal to noise ratio becomes too low, there is no warning, it just stops working. Analogue systems are less susceptible to this problem, they break up more gradually, the SNR becomes lower and lower, but our ears are good at filtering out some of the problems and it serves as a warning to the artists and their staff to fix the problem.

In the following graph an operating point (blue dashed line) is marked, where digital outperforms analogue in terms of quality. Analogue transmission shows degraded sound quality because of worse RF conditions. Digital transmission however can keep its sound quality.

The operating point is dangerous, as it is very close to the digital cliff. A few dB more of link attenuation like putting fingers around microphone antenna would lead to total loss of digital link. An analogue link would still work however slightly degraded. Analogue transmission shows graceful degradation as long the operating point is kept above FM threshold, which we may call the analogue cliff.

With FM modulation in analogue PMSE 1 dB loss in RF SNR directly turns into 1 dB loss in Audio SNR. There is a linear relation above FM threshold.

Passing the digital cliff leads to a fully audible dropout, whereas the graceful degradation mechanism in analogue is barely audible for untrained ears. Audio Technicians will hear this and take appropriate actions like instructing the artist via IEM not to put fingers around microphone antenna.
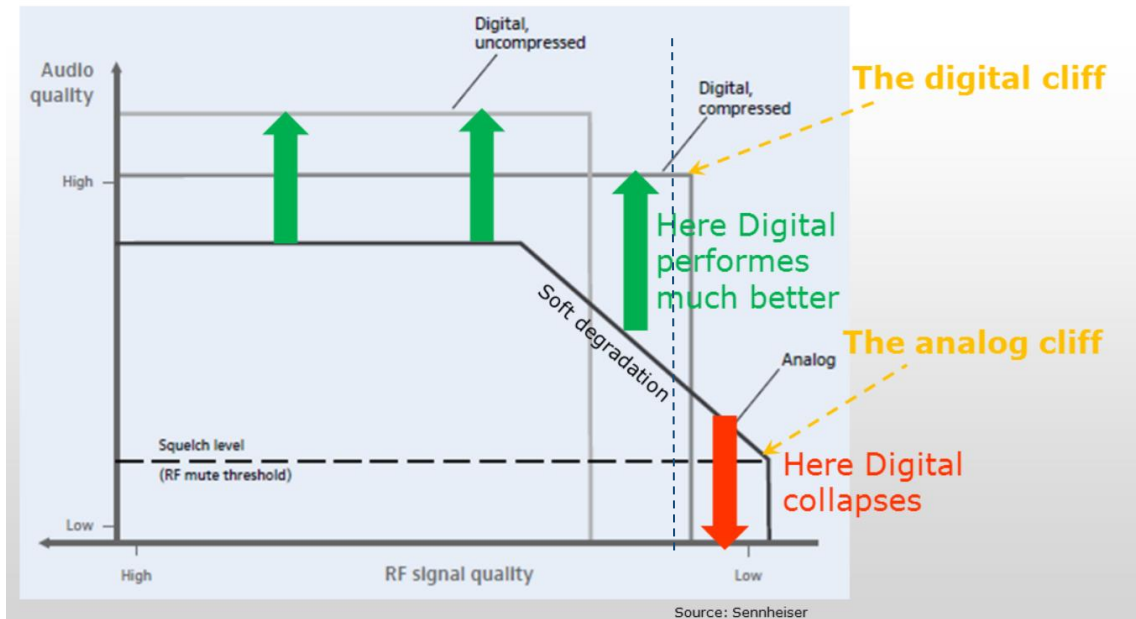

Figure 16 : Analogue versus Digital Cliff

However digital has some properties that make it more attractive before the digital cliff hits and that make it hard to recover from failure. In a digital signal the overhead necessary to transmit and receive the data makes it less susceptible to interference. The overhead gives information on the number of bits sent and therefore received. It may contain certain forms of error correction or other ways to derive a conclusion about the correct signal. Analogue contains no such data. Therefore as long as there is a good enough signal, even after soft degradation has set in in analogue systems, digital systems can still deliver high quality audio. Indeed there appears to be some debate between professionals whether or not digital can work with lower SNR-levels compared to analogue. Though theoretically this may not be the case knowing that a signal can be only 0 or 1 and knowing the control data actually allows the system to operate 'better' at lower signal to noise levels, though at a higher risk from dropping of the digital cliff.

Once digital has lost the link a significant time is necessary to retrain the system to the correct state. It will take roughly 500ms or half a second before the system has again enough data to correctly separate control data and audio to retrain. Given that in some analysis 80ms is already considered a pause in some languages/linguistic contexts and 250ms is internationally in academia considered a pause [6] 500ms is a significant and very noticeable pause. Recovering from that is hard for an artist. In a speech it might be possible

for a speaker, but it shouldn't happen too often or both the audience and the speaker will find it difficult to deal with.

A common mitigation technique, particularly in digital wireless audio is not to use one wireless link, but to use 2 or even 3 spectrally diverse digital wireless links to send the signal over. This way even if the main signal cuts out for a reason (range, interference) there is good chance the other signals can still continue. This means that instead of sending on one frequency the microphone or the PA system is now sending on multiple frequencies and the spectrum budget decreases accordingly.

A difficulty that was mentioned a few times in interviews is that for interference the average signal strength is not relevant, it is the peak and its duration and how long it interferes with the signal. So a piece of spectrum may look clean on a scope at one resolution but here may be spike at short time intervals, this renders the spectrum unusable particularly for latency critical digital transmission as it may just hit the retrain buffer of the signal, resulting in a much longer signal loss than on analogue, where it may be only a minor pop, something the ear can work around and the audience may not hear in the combination of instruments, singing and noise from the audience.

5.3 Bounds on dense packing of microphone frequencies

Wireless devices have a spectrum mask that defines how tight the devices can be packed together in a band. Below the spectrum masks defined in the relevant Standard EN 300422 (*Electromagnetic compatibility and Radio spectrum Matters (ERM); Wireless microphones in the 25 MHz to 3 GHz frequency range; Part 1: Technical characteristics and methods of measurement*) are given for digital systems below and above 1 GHz. (Figure 17 and 18)
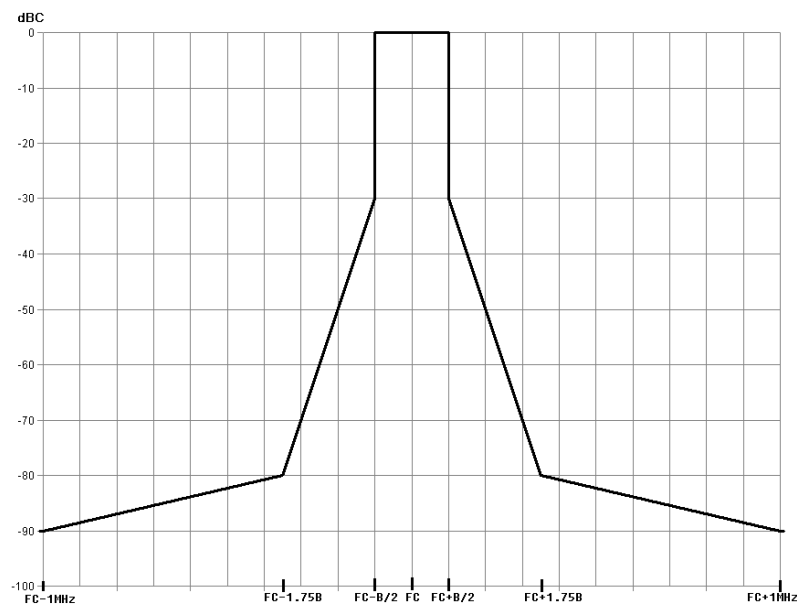

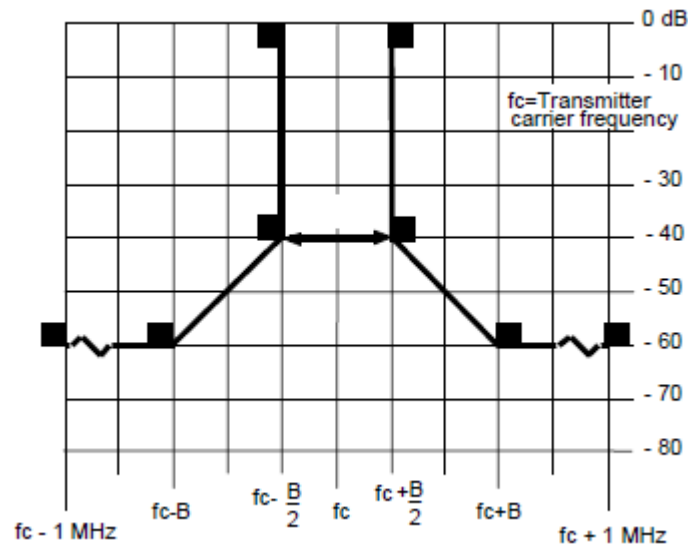**Figure 17 :** Spectrum mask for digital systems below 1 GHz

Figure 18 : Spectral mask for digital systems above 1 GHz, normalised to channel bandwidth B

For the following analysis a declared bandwidth of 200 kHz is assumed. For the case below 1 GHz the second knee is found about 250 kHz spacing from the edge of the occupied channel. For the case above 1 GHz, 200 kHz spacing can be derived.

Below 1 GHz at the second knee the signal is 80 dB down, whereas for the case of above 1 GHz the signal is 60 dB down. Now as a second assumption let's assume that these rejection values are sufficient to handle the near-far problem, meaning that one mic could be near and one far from a receiver. There is a risk that the wideband noise by the mic that is located near to the receiver could shade the weak signal by the mic that is far from the receiver. In essence the mask defines the maximum tolerated level difference between near and far mic to the receiver.

If 40 dB SNR is needed min, than a difference in propagation loss of about 30 dB can only be tolerated with a spectral mask 70 dB down, as otherwise the side emissions will start to dominate. A leading supplier of PMSE equipment has stated that 23 PMSE frequencies would be possible for standard (High Quality) mode inside 8 MHz. This accounts for roughly 350 kHz equal spacing of mic frequencies.

The following analysis shows that this reflects the maximum possible given the spectrum masks in EN 300422.
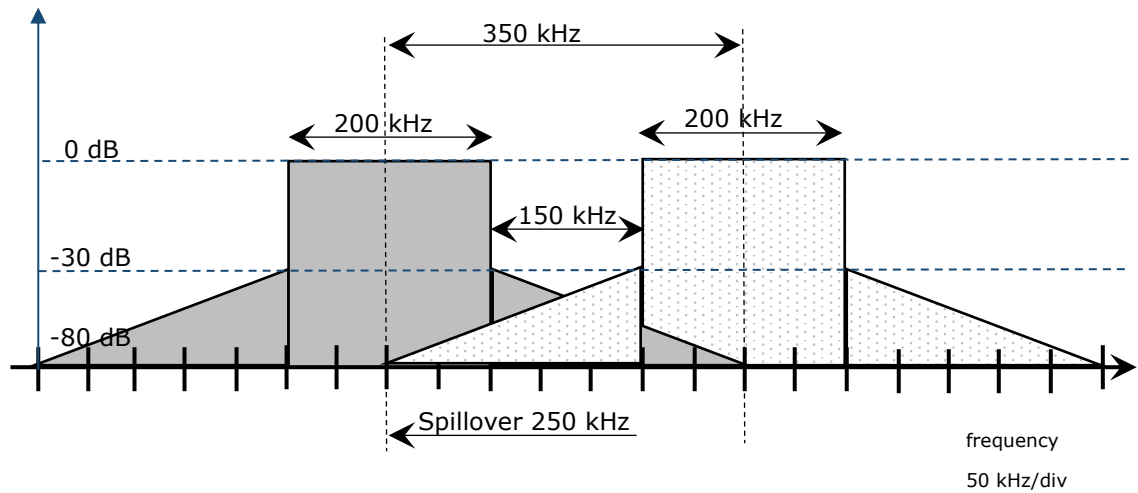
Figure 19 : Overlap of spectrum mask with today's equidistant spacing of 23 Ch/8MHz for spectrum <1GHz
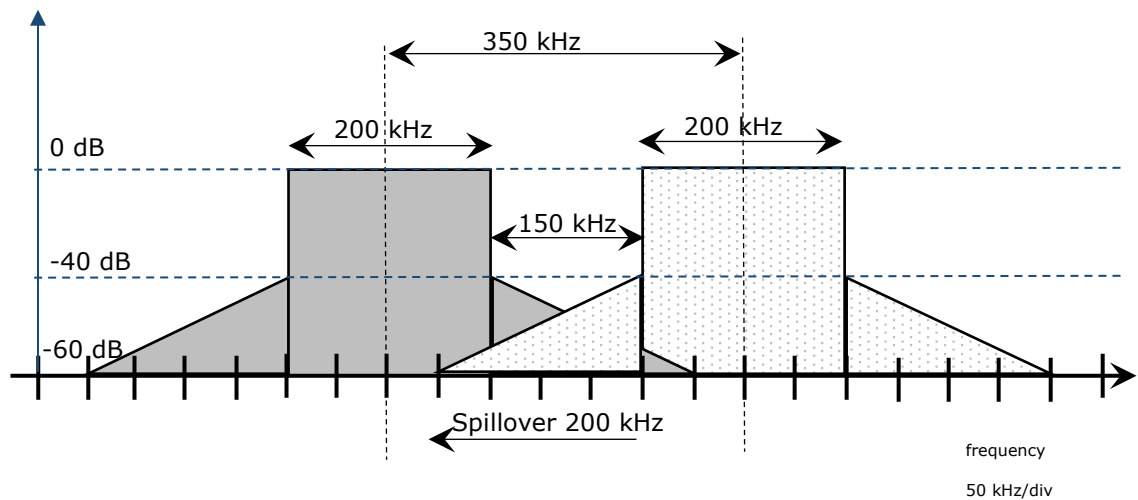

Figure 20 : Overlap of spectrum mask with today's equidistant spacing of 23 Ch/8MHz for spectrum >1GHz

Figures 19 and 20 show that there is not much possibility to pack microphone frequencies more dense based on today's spectrum mask for a 200 kHz microphone channel. There is already "spill over" from neighboured microphone frequencies. More spill over, which would arise from more dense packing cannot be tolerated as this would reduce dynamic range. In real practical scenarios a PMSE receivers faces the so-called "Near-far Problem". This means one mic may be close to the receiver and one mic maybe far away. There is a risk that the spill over form nearby microphone gets stronger than the weak signal from a mic far away to be received as wanted signal. Level variation by near far may easily account for 50 dB, so it gets on the order of the spill over. In summary it has to be concluded that the PMSE industry

has already worked intensively to make above scenario happen ensuring high quality and low latency audio already with 350 kHz equidistant placing of mic frequencies.

If spectrum mask would be tightened a more dense placing would become feasible. However that is technologically ultimately challenging. A spill over by the width of the wanted channel (200 kHz) is more or less natural as it is a consequence of always present non-linearities in RF circuitry. The spectrum mask for spill over in PMSE starts about -30…-40 dB and goes down to -80…-60 dB. That already is very demanding. In cellular communications neighbour channel power is specified about -31…37 dBc for LTE terminals summed over the complete neighbour channel, which is less demanding compared to the PMSE case here. Once again the PMSE manufacturers have already provided very clean transmitters with a spill over less compared to cellular. It is even the case that the spill over/adjacent channel leakage allowed by spectral mask for PMSE is already tighter than for cellular base stations (-45 dBc).

In 2013 Cambridge Consultants in a report for Ofcom mentioned the possibility to move in the future to 350 kHz spacing for digital and 400 kHz spacing for analogue at 2:1 compression. This would allow in theory up to 3 wireless channels per MHz and therefore up to 23-24 devices per channel for digital and 20 for analogue, if careful planning is taken into consideration.

The following conclusions can be derived then in ideal situations.

Table 4 : Packing of frequencies

	< 1 GHz	> 1 GHz
Assumed channel bandwidth	200 kHz	200 kHz
Spacing/guard band derived from modulation mask	250 kHz	200 kHz
Packing from mask: no of channels in 8 MHz	17	20
Today's Packing: High quality mode in 8 MHz	23	
Today's Packing: reduced quality mode in 8 MHz	63	

The reduced quality mode allows Shure to pack 63 channels in 8MHz. The constraints of this system are however that the transmission power has to be brought back to 1 mW. This reduces range and requires a very low noise environment. It also limits how microphones can be used. It requires the devices to be stationary, a little bit spread out, but not too spread out. It is therefore usable in conference situations, but wouldn't deliver the required quality for a full choir. Other manufacturers have similar high density modes using both digital and analogue techniques, but they are on the edge of what is possible.

From Table 4 it is clear that already today the limits by the spectrum mask are fully explored. The leakage into neighboured PMSE channel is explored by min C/N required plus a margin to accommodate for the near far problem. Thus it cannot be deduced that there will be dramatic gains in the future. Gains could only be envisioned for the future if the spectral mask would be defined in a steeper way.

Today's dense packing is facilitated by isolators. Spending more effort through a double stage isolator would not provide higher gains in packing. Nevertheless further research on isolators is needed to overcome the narrowband behaviour of magnetic isolators. Electronic

(magnetic free) isolators could offer frequency agility. In paragraph 5.4.2 the research on isolators/circulators is discussed in more detail.

5.4 Intermodulation

So far this report has mostly focused on single links. However the goal of this report is to research the effect of digital wireless in large scale PMSE events. It is in the large scale that real problems exist. A transmitter will pick up neighbouring signals and rebroadcast these. The more signals there are, the more they are retransmitted. In addition to the original signal there is also interference between the intended transmission and the rebroadcasted signal that was picked up from other transmitters. This is known as transmitter intermodulation. (Figure 21)
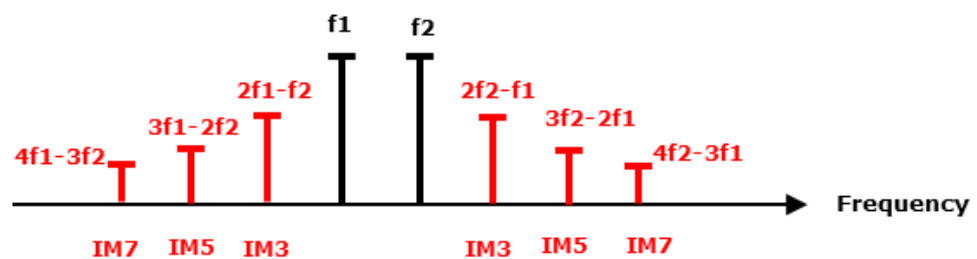

Figure 21: Transmitter intermodulation

Particularly difficult with transmitter intermodulation is that it works progressively. The more transmitters there are, the more intermodulation they create in neighbouring frequencies, which in turn adds to the noise floor and makes those frequencies unusable. The graph below shows that in order to run intermodulation free only 47 microphones would fit in 140 MHz of spectrum (Figure 22). This would be roughly 1 device per 3 MHz. However in practice CEPT [1]reported that more than double that could be reached, because of spatial separation, different transmitter characteristics, better hardware, power etc. This would equate to 1 per 1.5 MHz. According to interviewees the industry standard is already at 2 per MHz, or between 14 and 16 devices per 8 MHz, if enough care is taken to diminish intermodulation products.

As one interviewee said; I don't care about interference either from the noise floor, or from active interference through intermodulation or other transmitters nearby, but whether I can get a workable signal from it (and that is easier with a low noise floor and no other active equipment).
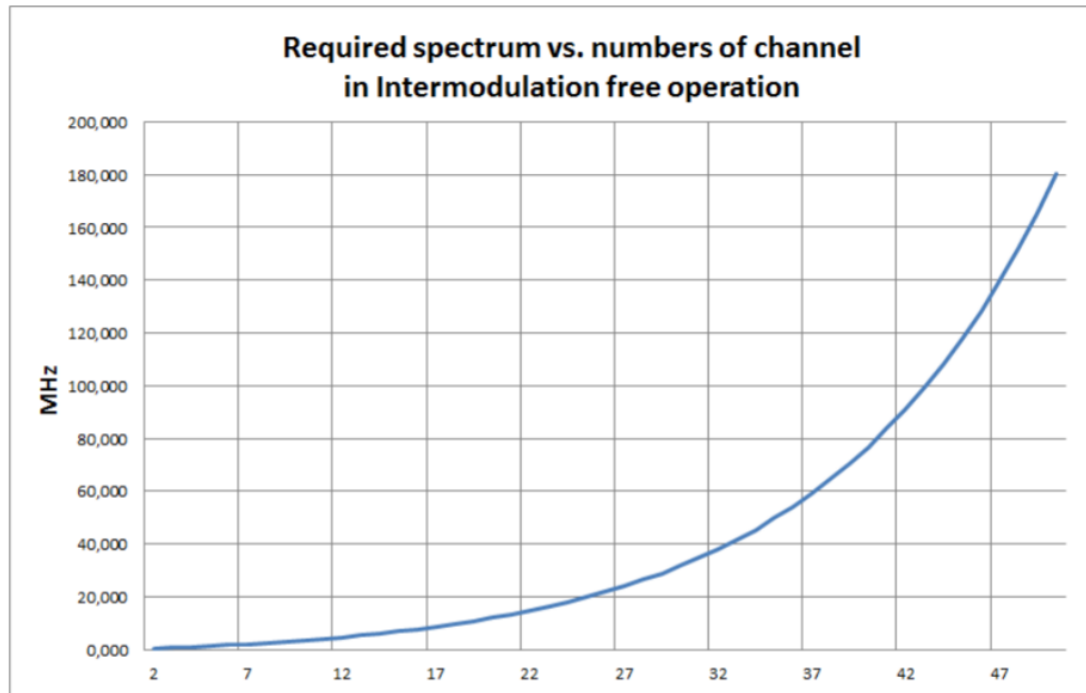
Figure 22: Required spectrum vs. number of channels in 'intermodulation free operation' [CEPT]

CEPT looked into this problem and came up with four mitigation scenarios.

1. Frequency planning, in order to avoid intermodulation products which create interference on useful signal;

2. Integration of output filters and/or ferrite isolators;

3. Control of microphones transmitted power;

4. Adoption of transmission technologies that support operation in higher interference environment.

CEPT concluded that analogue systems support the first two mitigation techniques. Cognitive systems support the first three mitigation techniques. Digital systems support all four mitigation techniques. If all 4 are used the limit to the packing of mic frequencies is not the intermodulation product, but the spectral mask.[18]

In interviews it became clear that digital wireless microphones are much more vulnerable to the effects of transmitter intermodulation. If other transmitters start emitting 0 and 1s at the same frequency as where a receiver is listening, this may cancel out these bits and make them unheard or even flip them from 0 to 1 and vice versa, causing bit errors. As a result it becomes unusable. However in practice this has turned out to become the basis of their strength, as their designers design them to be very good at handling IM-products.

---

[18] Spectrum planners don't fully accept that the four mitigation scenarios don't apply to analogue systems. Particularly control over transmitted power is one that is not actively done during an event, but before the event and therefore can apply to analogue devices too.

5.4.1 Frequency planning

Frequency planning has become integral to high end PMSE planning. At larger events there are dedicated frequency planners whose job it is that intermodulation products do not exceed the SNR necessary for other devices to operate in. They use spectrum analysers and software to determine whether there are any IM products present and whether a new device will add unacceptable IM products if it is turned on. The frequency planner for the Rio 2016 Olympics gave 3 ways to deal with IM products.

These processes allow for a more efficient use of the spectrum; some of them are outlined below, with advantages:

Spectral segregation	Utilising different frequencies in the same physical location at any one time	The most common and reliable form of segregation. The ability to monitor each carrier independently allows security. Carriers not part of an RF solution can easily be detected.
Spatial segregation	Utilising the same frequencies in different physical locations at any one time	Used most often for solutions that have high numbers of carriers over a large physical area. Cross-contamination of segregated areas more difficult to detect.
Temporal segregation	Utilising the same frequencies, in the same physical location at different times	The least reliable of all three methods, as it requires carriers to be turned on or off at pre-defined times. There is no 2nd level of safety for this form. Contamination is a high risk.

* Standard intermodulation calculations are used in conjunction with the above weightings

The three different types of segregations above carry various factors of reliability, and can be used separately, or together with one-another, depending on the circumstances. Each of the IM products is calculated evaluating frequency, time and position for each device, both receiver and transmitter according to the characteristics and tolerances of that device. This is how 270 frequencies just for the ceremonies alone were planned, after which additional frequencies had to be planned for the broadcasters that were present. [19] When actually operating a new nearby TETRA installation, for emergency and security services, also caused some IM products that hadn't been foreseen and had to be dealt with in cooperation with the operator of that installation.

---

[19] http://soundhub.audio/rf-coordination-rio-2016-steve-caldwell/

5.4.2 Integration of ferrite isolators and output filters.

Another way of dealing with intermodulation mentioned by CEPT is to make sure that the transmitter is less susceptible to it. What doesn't get picked up or what isn't rebroadcasted can't be a problem anymore. Circulators/isolators play a key role in improving transmitter intermodulation performance. They prevent RF signals traveling backwards into a PMSE transmitter. An isolator is a two port directional line, which is realised when the third port of circulator is terminated in a matched resistive load (Figure 23).
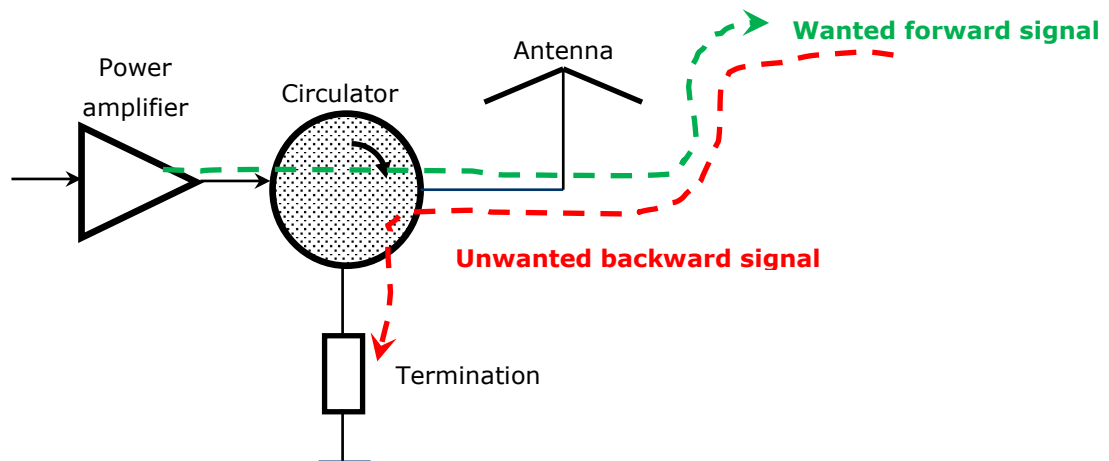

Figure 23 Typical circulator/isolator arrangement with transmitter

Circulators/isolators are non-reciprocal passive devices. They are realised based on anisotropic material like YIG ferrites (Yttrium Granite). The power of signal getting backwards into the transmitter is turned simply into heat at the termination resistor. In some systems there are multiple circulators/isolators in sequence, for example in cellular base stations often double stage isolators are used to obtain very high backwards attenuation. It appears that for PMSE applications a single isolator per band is good enough, but multiple could be implemented.

As a very wide tuning range of e.g. an octave is desired with PMSE devices, one wideband isolator cannot provide sufficient performance. Instead a bank of narrowband ones (e.g. each 20 MHz) and two switch matrices are implemented (Figure 24). There is a large cost hit by this approach and much PCB area is occupied, which make isolators mainly showing up on premium products and not at low end ones. However, improving TX intermodulation allows for more close spacing of PMSE links along frequency axis. Thus a more efficient spectrum use could be achieved if every PMSE transmitter would be equipped with one. The necessity for IM control is mainly a consequence of the high operational SINR PMSE links require.
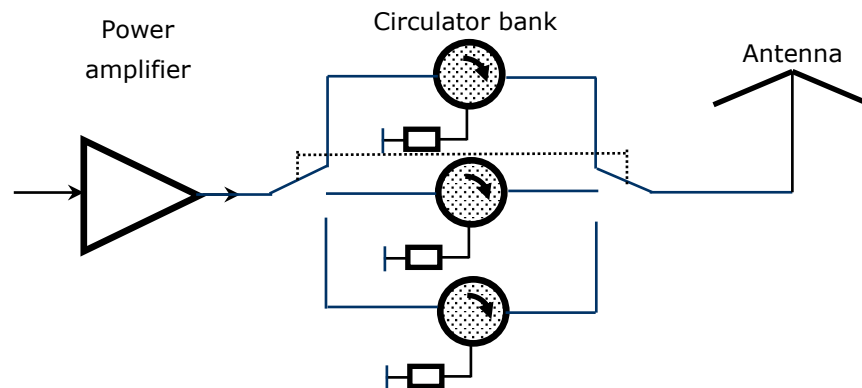
Figure 24 : Wideband approach by circulator ban

And so in order for digital wireless not to suffer from bit errors in transmission that are the result of intermodulation, it comes with a large number of very good circulators. The isolator presents a constant termination impedance to the RF power amplifier ensuring its linearity even if the antenna is detuned by placing it closer and fare to the body. Forward linearity of RF power amplifier is influenced by termination impedance. This is uncritical on analogue FM system that operate constant envelope.

Isolators could also be used widely with analogue wireless microphones. However it is less economical to do so. Due to price competition and tuning range demand with analogue mics typically there is no isolator. As a result the balance in cost effectiveness from the perspective of the manufacturer and users is different. For the end-user the circulators add cost, but no increase in audio quality and limited mitigation of intermodulation effects beyond what a spectrum planner can do already. So their main benefit lies in more close spacing of devices. For digital wireless microphones however they enable the system to exist, deal with intermodulation, and a more close spacing. So the relative benefit is higher and that explains why isolators despite being a significant cost factor are more accepted on digital wireless systems. There are however high-end brands that do sell analogue systems, that have circulators on board. The manufacturer markets these for situations where a large number of wireless devices are used in parallel and where a wide band of up to 320MHz can be used. Most wireless systems can only operate in a limited range ie 40 MHz. As a result these wideband systems are a good fit for large scale and/or travelling use.

New academic research on circulators

Very recently new research on circulators has opened up some possibilities to design circulators that are designed in silicon instead of using YIG ferrites. These are known as "electronic circulators", sometimes also called "magnetic free circulators" [7].The key idea here is that a nonreciprocal behaviour is achieved through a fast switching section (Figure 25 and 26).
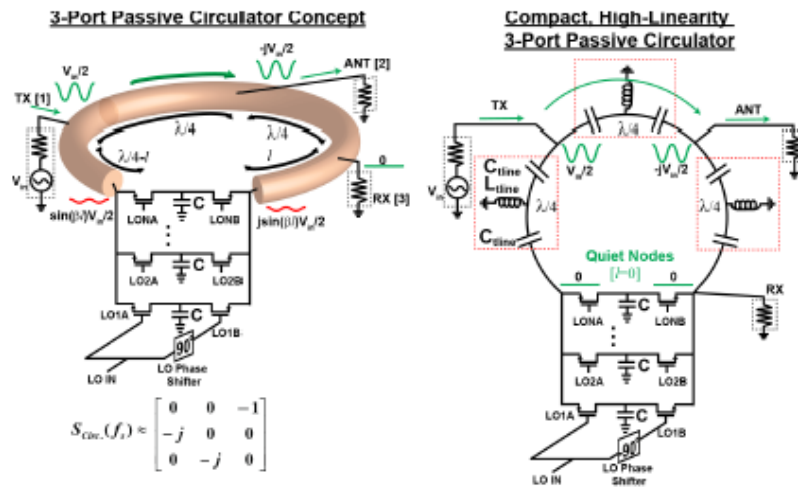
Figure 25 : Electronic circulator

Fast switching is ultimately compatible with low cost RF CMOS technology. Thus the electronic circulator can be integrated on chip. The problem of cost and form factor is gone by this. The behaviour is narrowband, however electronic circulators can also be tuned and reconfigured electronically in situ for their frequency of operation.
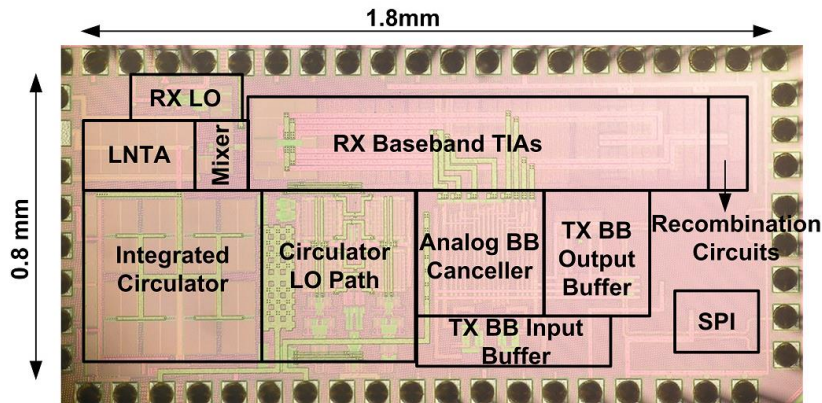

Figure 26 : Electronic circulator integrated on chip

Electronic circulators can serve as a low cost way to bring also isolator technology into low end products and thus into widespread use. Such an improvement of transmitter hardware would allow for more efficient spectrum use at low cost. However further research and development effort is needed in this area to bring it to wide commercial use.

5.4.3 Control of transmitted power

CEPT lists control of transmitted power as a benefit of cognitive systems. In cognitive systems there is a feedback from receiver to transmitter on what power level still gets an acceptable signal. This allows the sender to reduce the power thereby reducing the IM-products. If TX power is reduced by factor 2 , so -3 dB, then IM product fall faster. An IM3 product falls by -9 dB (3x), an IM5 product falls by -15 dB, an IM7 product falls by -21 dB. Such differences are really large particularly in situations where there are dozens if not

hundreds of devices transmitting. It does however add significantly to the complexity of wireless microphones, IEMs, not to mention the number of active links in already scarce spectrum.

To some extent however this technique is also used in large scale PMSE events for analogue systems, through the spatial distribution of devices. Not all 200 active links need to be in the same 100m^2 Often they are distributed over a number of fields and tents. By moving them and the receiving antennas around spatially, their IM products decrease as well as these are picked up and retransmitted less. However if the IM problem is managed the spectral mask problem still persists.

Many interviewees suggested that in general digital systems can work better with lower TX power than analogue systems, because the transmitted signal is better known by the receiver through the control data. The fundamental characteristics of digital transmission being either 0 or 1 also helps according to some, as it is more clear what needs to be detected. So even if there is no active control layer, the control of a good spectrum planner can greatly reduce the power needed and therefore the IM products. However this is contested by others who work with high-end analogue systems.

5.4.4 Adoption of techniques to deal with interference

The adoption of techniques to deal with IM is what underlies particular high density systems for example Shure's Axient system that allows the use of 63 devices in an 8MHz channel. Now this is impressive, but in practice in comes down to combining all elements of the previous paragraphs. The combination of strong compression, with excellent hardware, low transmission power and low noise floors allows this number of devices. However this goes hand in hand with reduced audio quality, reduced robustness, latency issues etc. Therefore it is of little use in large scale PMSE events.

In a report by Cambridge Consultants it was identified that Wisycom's WTH400 analogue system is another example of a system that deals with intermodulation through the use of the spatial distribution to use two identical power amplifiers in phase quadrature, combining them in a phase-shift network at the antenna output. This causes any signal received at the antenna to cancel. The disadvantage is that the power amplifiers have to be closely matched (typically requiring an expensive calibration process in manufacture) as well as the additional component cost and power consumption. This does allow them to pack as many as 30 devices in a channel because of the cancellation of IM products.

The resulting improvement in back intermodulation performance is about 25dB compared to a single PA, and with care in deployment allows up to 30 audio channels in 8MHz. This care includes ensuring that body-worn transmitters are in place on the person before switching on, as the 25dB improvement can be rapidly wiped out by the much higher signal level that can result from a transmitter antenna which has no associated body absorption.

The conclusion therefore is that fundamentally it isn't the digital or analogue nature of the transmission that deals with the intermodulation products. The solution lies in the quality of the hardware used and the quality of the spectrum planner who plans the event.

5.5 Wideband PMSE systems

Robustness in transmission is achieved through diversity. Making use of Wideband PMSE might solve this problem. Such diversity could also be a solution for the intermodulation problems that are the basis for many of the problems in PMSE. Diversity comes in different flavours (Figure 27)[20]:

1) Antenna diversity

   a) Spatial diversity. This comes through multiple antenna locations

   b) Polarisation diversity. This comes multiple antenna polarisations at one location (see for example the Wysicom system mentioned earlier)

   c) Pattern diversity. This comes by using different antenna types that having different radiation patterns

2) Temporal diversity. This comes through channel coding, block codes and interleaving in digital transmission

3) Frequency diversity. This comes by frequency hopping in narrowband systems, e.g. implemented in 2G GSM or by wideband interfaces, e.g. implemented in W-CDMA/UMTS or LTE.
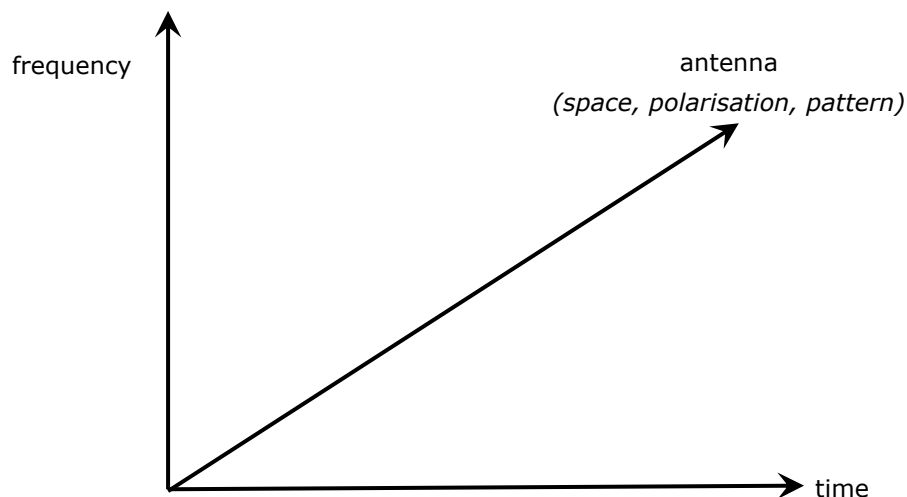

Figure 27 : Diversity Options

Ad 1)

With PMSE spatial diversity and its variants polarisation and pattern diversity is explored through multiple antennas on the receiver side. At transmit side typically one antenna is used. This could be extended by migrating to MIMO e.g. a 2x2 system meaning two antennas on transmit and two antennas on receive side. As antenna diversity can also be realised through polarisation or pattern diversity large spatial separation is no must to make MIMO work. So MIMO is not contrary to compactness.

Experience further tells that high correlation factors and bad isolation between antennas does not compromise antenna diversity gains a lot. Thus it is worth implementing even under constraints of compactness as PMSE has.

---

[20] See also ETSI TR 103 450 V1.1.1 (2017-07) System Reference document (SRdoc); Technical characteristics and parameters for Wireless Multichannel Audio Systems (WMAS)

Ad 2)

Temporal diversity cannot be explored in PMSE systems due to latency constraint. Most digital systems in use so far have less stringent latency requirements. They are gaining their robustness mostly from coding and interleaving over times frames on the order of 40 ms (GSM) or more. Latency than equals minimum twice the interleaving time as interleaving on TX side has to be de-interleaved on RX side. Thus total latency would be 80 ms for above GSM example.

Temporal diversity gets beneficial if the block length is longer than the coherence time of the channel. If the coherence time is 5 ms (typical value), than interleaving is of no use when looking for block length of 1 ms as in PMSE.

ad 3)

Frequency hopping is not in use with today's PMSE. PMSE uses a static narrowband frequency assignment. In 2G GSM frequency hopping is done within a radio block, so the channel coding explores the frequency diversity gain. However long block length are contrary to short latency.

Frequency diversity has a further charming side effect. While hopping around in frequency domain also different interference situations are faced. When using long block length in channel coding the interference conditions at different frequencies are somehow averaged. Once again this cannot be explored in PMSE due to stringent latency requirements. Therefore some people say frequency diversity comes along with interference diversity. But this only is present when long block length are used.

Frequency diversity is in place if the bandwidth explored for transmission is wider than the coherence bandwidth, which is typical on the order of a few MHz. That is why 3G UMTS with 5 MHz channels is a wideband system. 2G GSM and PMSE with 200 kHz are thus narrowband systems.

5.6 What are the possibilities for further develop PMSE?

Benefits from digitisation of PMSE are not that high as not all degrees of diversity present in other digital systems can be explored due to the stringent latency requirements of PMSE.

- It can be said that antenna diversity and MIMO techniques could be explored in PMSE. There are opportunities that could be grabbed in future to improve robustness of transmission.
- Today PMSE is a narrowband system with 200 kHz channel bandwidth. This is a significant drawback and makes PMSE vulnerable to frequency selective fading. In order to overcome this PMSE could migrate to wideband air interfaces like OFDM. This would allow to pick frequency diversity gains without paying the price in latency which is the case on frequency hopping.

Wideband air interfaces like OFDM are very complex and a new development would consume a lot of time and effort to turn into effect. From that perspective it looks attractive whether

already existing wideband air interfaces could be taken over for PMSE. This is the reason why the PMSE-xG project in Germany studies whether 4G and 5G air interfaces would be suited for PMSE. It would also allow the PMSE manufacturers to take existing MODEM solutions from the market and integrating these into their PMSE devices, thus avoiding development of an own wideband air interface technology. Qualcomm e.g. has advertised OFDMA technology for PMSE. [21]

One might say that modem solutions for OFDM are too complex and power demanding. This does not fully hold if COFDM (Coded OFDM) is used, when information from one device is mapped on selected subcarriers in an OFDM signal, that have a wide frequency spacing that is larger than coherence bandwidth, so that still frequency diversity is explored.

If PMSE would use a wideband air interface, the spectral mask problem would be gone. Also the intermod problem would be gone. All margins due to frequency selective fading would be gone. Most of today's problems are due to the narrowband nature of PMSE with 200 kHz channel bandwidth, which applies to both analogue and digital systems. Digitisation in PMSE does not address this narrowband nature of PMSE. This is of historic reasons, as PMSE started from being similar to FM Radio technology. Indeed stereo IEMs use the same stereo chips as FM radios.

Regarding costs it is key that highly integrated microelectronic circuits are available on the market. As PMSE does not reach quantities as smartphones the PMSE manufacturers barely can afford their own chip developments. It is therefore advisable to take over chips from other markets like from cellular 4G/5G. An own wideband air interface just for PMSE would persistently struggle from high costs for integrated chips due to low quantity. And indeed this is a further motivation behind PMSE-xG project. The number one objective in microelectronics is quantity to bring costs down.

If we look back we see that PMSE took over FM radio technology. Now PMSE could consider taking over cellular technology if requirements can be met. The 2G (GSM) and 3G (W-CDMA) technologies could not likely be used to meet the PMSE requirements, but 5G technology is more likely to meet the latency requirements of PMSE. Although 4G technology cannot serve the latency requirements for professional PMSE use but it might be sufficient for certain market segments.

Wideband PMSE might therefore be a worthwhile area to research, however as with any new radio technology mass implementation is likely to be more than a decade away.

5.7 Conclusion on radio aspects

The conclusion of this segment is that digital radio may enable some improvement over analogue systems when comparing transmission. It is however not so clear that this is because of digital as the modulation scheme for transmission. Some manufacturers can use analogue PMSE-equipment to achieve very high channel densities with low intermodulation products to allow for equal distance spacing. The products of these manufacturers appear to

---

[21] http://www.marcus-spectrum.com/Blog/files/b6610ee4fe16b2c5effe788cbe718840-155.html

be on par with digital transmission systems manufactured by others. It is therefore unclear a comparison between these systems is a comparison of analogue vs digital or whether it is a comparison of manufacturer A vs manufacturer B. Discussions with the industry have further shown that despite the headline grabbing device counts per channel of some manufacturers other manufacturers focus on other elements, such as quality and robustness. Indeed, both digital and analogue systems still are manufactured and sold to high end customers.

There are some indications that digital systems are more susceptible to bit errors due to intermodulation. As a result manufacturers have included isolators/circulators in their digital products. [22]Manufacturers had to invest in better hardware at a cost to the user too, as these systems are less flexible and more expensive. However it has allowed the use equal channels spacing and more devices in a band, without significant interference from intermodulation products of devices in adjacent channels. This is not perfect, there will always be a transmitter intermodulation, however digital allows this to be less pronounced and therefore more workable for the production. This will help smaller scale events more than bigger scale events. It becomes easier to scale from 20 to 60 devices, but the C-level planners already mitigated many radio problems. For them the better hardware will make some elements easier, but it doesn't translate in a dramatically higher number of active devices in a band or across an event that uses multiple bands. Now that less bands become available, this only decreases the total number of links that can be achieved for an event.

---

[22] Intuitively, Moore's law should provide for higher data rates at a lower (frequency) budget as wireless communication methods such as WiFi and mobile communication gets faster and faster. But Moore's law is only partly responsible for this. Other factors that also contribute to this trend are the use of more and higher frequency bands, smaller and denser (mobile) cells and shorter distances, use of latency to apply buffering and smart compression schemes. In the PMSE sectors it is much harder to improve the use of these factors.

6   Scenarios

This report aims to investigate the potential difference between digital and analogue wireless links in PMSE applications. Fundamentally there appears to be no considerable gain in the use of digital technology. In audio encoding analogue has an advantage that it doesn't need a lot of overhead to transmit the signal, whereas digital has as the advantage that because it has the overhead it can be more predictable in transmission. However analogue systems use companders, which are less consistent in their performance, compared to digital. On balance digital systems are on par or slightly outdo analogue systems when it comes to audio quality. It is therefore likely that digital systems will be the majority of systems deployed in the future. However there is no fundamental reason to assume that given the same level of engineering and costs, analogue and digital systems are not roughly on par with each other.

We therefore estimate that currently and in the near future the following spectrum demands will be realistic based on lower and higher counts of number of wireless links. The lower count here reflects the situation where audio quality and resilience are emphasised more, or where planning is harder because more noise and interference has to be dealt with. The high end represents the situation where state-of the art equipment, good spectrum planning, consistent application, good RF shielding indoor, low noise floor etc. all play a role.

Event type	Wireless devices 2024	No of channels Low end (16/8MHz)	No of channels high end (24/8MHz)
Television	140	9	6
Outdoor Music Festival	260	16	11
Theatre	94	6	4
Sports	140	9	6
Film making	200	13	9
Television news	45	3	2
The Hague election	160	10	7

6.1 Scenario analysis of large events

Below two main scenarios for large events are described. One for a multistage scenario with multiple stages with each a number of PMSE devices in use, and one for a compact event with a large number of PMSE devices on a single stage.

6.1.1 Multistage scenario

With a large event, where multiple stages are involved, spectrum can be saved if frequencies are reused. Of course this can only be done if locations where identical frequencies are used have a large enough spatial distance so to provide sufficient attenuation by propagation. In essence the attenuation from propagation has to be sufficiently large that Carrier to Noise

C/N requirements are met. An example could e.g. be a large sports event where reporting about different disciplines is happening at different places widely spread apart. Another example could be large music festivals with bands at different stages. In the following it is analysed how a reduction in C/N requirement allows for tighter reuse or closer distance for frequency reuse.

From link budget planning the following calculations are known. Receive level decays with distance r with n being the pathloss exponent. As this is a reciprocal scenario the pathloss exponents for C and N are assumed equal[23]:

$$P_{RX} = c \cdot \frac{1}{r^n}$$

$$P_{RX,dB} = c - 10 \cdot \log_{10}\left(r^n\right)$$

$$P_{RX,dB,1} - P_{RX,dB,2} = \left[c - 10 \cdot \log_{10}\left(r_1^n\right)\right] - \left[c - 10 \cdot \log_{10}\left(r_2^n\right)\right]$$

$$\frac{\Delta P_{RX,dB}}{10} = \log_{10}\left(\frac{r_2^n}{r_1^n}\right)$$

$$\frac{\Delta P_{RX,dB}}{10} = \log_{10}\left(\frac{r_2}{r_1}\right)^n$$

$$\frac{\Delta P_{RX,dB}}{10} = n \cdot \log_{10}\left(\frac{r_2}{r_1}\right)$$

$$\frac{\Delta P_{RX,dB}}{n \cdot 10} = \log_{10}\left(\frac{r_2}{r_1}\right)$$

$$\frac{r_2}{r_1} = 10^{\left(\frac{\Delta P_{RX,dB}}{n \cdot 10}\right)}$$

Now assuming that digital PMSE could do with 10 dB less C/N as OFCOM stated, this can be converted to a radius change. However a reasonable pathloss exponent has to be assumed for this conversion. In Cellular network planning at UHF bands a pathloss exponent of 3.4 is assumed based on Urban Okumura Hata Model as depicted in Annex B.1.1 of ETSI GSM 03.30.[24] A pathloss exponent of 2 would reflect free space, but is too optimistic as it reflects the case of no objects and free space propagation. In real scenarios signals decay faster with

---

[23] We have assumed the two stages using identical technology. Either both stages use analog or both stages use digital technology. However if technology mix is present, so that one stage uses analog and one uses digital technology one can argue that such a mixed case is dominated by the worst performing technology. So if analogue PMSE requires higher C/N, thus a stronger RX level, then the mixed cased can be treated as if both stages were equipped with analogue. If there is outage it does not matter whether it happens at both stages or at one stage only..

[24]GSM 03.30, TR 101 362, Digital cellular telecommunications system (Phase 2+); Radio network planning aspects (GSM 03.30 version 8.3.0 Release 1999)

distance, so pathloss exponent is larger. This helps in isolating stages and reusing frequency at closer distance.[25]Inserting these values delivers:

$$\frac{r_2}{r_1} = 10^{\left(\frac{-10dB}{3.4 \cdot 10}\right)} = 10^{\left(\frac{-10}{34}\right)} = 10^{-0.294} = 0.512$$

This can be interpreted the following way: If C/N requirement is relaxed by 10 dB thanks to migrating from analogue to digital as OFCOM has stated, under the assumption of a pathloss exponent of 3.4 this would allow to reuse the frequency at half the original distance.

For the case that pathloss exponent is only 2, a factor 0.31 is obtained, which means that this would allow to reuse the frequency at one third of the original distance. As the above analysis is related to the interfering signal, worst case conditions occur at free space conditions. Therefore a reduction of reuse distance by a factor of three may be assumed, unless site specific situations lead to another conclusion[26]

However certain side conditions have to mentioned here questioning the benefit above. The C/N requirement reflects a minimum C/N for a target quality. As audio quality in digital PMSE incorporating source coding and analogue PMSE incorporating compander is difficult to compare as perception is involved and there do not exist objective audio quality measures, the C/N requirement for a target quality is somehow interpretable.

Furthermore it has to be taken into account that digital and analogue PMSE behaves differently when falling below the C/N requirement, e.g. due to a sporadic deep fade or an artist leaving the planes stage area unexpectedly. Digital PMSE will face the typical "Digital Cliff", which means the connection will totally drop. Analogue PMSE will offer smooth degradation, which means that for every dB below C/N requirement, also audio SNR will reduce by one dB. So analogue PMSE will not face a connection loss as digital does. A loss of connection in digital will lead to long drop-out times on the order of 100...500 ms, if resynchronisation has to be conducted, which is unacceptable.

For that reason frequency planners will either not take benefit from this 10 dB or use this 10 dB improvement for digital only in parts. If for example only a 3 dB C/N reduction is explored, this means a radius reduction by factor 1.2, which is not dramatic and does not lead to dramatic spectrum savings given that the location of stages is not a consequence of frequency planning. Typically, it is vice versa, that the locations of stages are given and a sufficient frequency plan has to be found. A radius reduction of factor 1.2 will then not turn into another more efficient frequency plan. This would mean that the C/N benefit from digital will not turn into a more efficient frequency plan with a large event.

6.1.2 Single stage scenario

With a large event, where there is only one stage, there is this question what benefits can be drawn from digitisation?

---

[25] A pathloss exponent of 2 means 20 dB more attenuation if distance is increased by factor 10 (20dB/decade). A pathloss exponent of 3.4 means 34 dB/decade.
[26] This analysis refers to relative distances. In the case of free space propagation the separation distance in absolute termers are larger than in the case of a path loss exponent of 34 dB/decade.

As elaborated above, the main benefit is from whether or not using isolators to relax the intermodulation problem. When using isolators the limiting factor is the spectral mask and not whether digital or analogue PMSE are used. The spectral mask however is identical for analogue and digital PMSE.

The methodology in the following therefore is how frequency plans look different whether or not isolators are used. This is done by using the SIFM Software[27], which is the Sennheiser Software for calculation of intermodulation-free radio frequencies for wireless microphone systems.

In the following, three cases are compared assuming an 8 MHz channel and a minimum spacing of 350 kHz:

- Frequency plan assuming some very weak intermodulation products thanks to isolators (noIM). This case serves as an upper bound. Isolators can also be placed in cascade to achieve such high IM rejection that they get irrelevant.
- Intermodulation only considering IM3 (IM3). If an IM level 50 dB below wanted signal is assumed, then also the audio SNR will be on similar order. However fading variation has to be taken into account. The tolerable IM level depends on the target audio quality.
- Intermodulation considering IM3 and IM5 (IM3_IM5)

Very experienced frequency planners sometimes also consider IM7 under extreme cases. This however is not supported in WSM software. The following results were obtained (Table 5). For reference also Shure's statement on their new offering "Axient digital" is included in the table.

Table 5 : Max number of IM free frequencies in 8 MHz, assuming min spacing of 350 kHz

	noIM	IM3	IM3_IM5	Shure Info for digital mics
Number of IM free channels	19	11	10	23

Shure can only achieve the 23 channels in 8 MHz by going to less than 350 kHz spacing, which will raise the risk of interference due to spectral mask as depicted above. Sufficient margin to the spectral mask given in dB is needed to account for near-far problem, but also for effects by body loss and body shadowing. Even if microphones stand still, one may face body loss and the other not. Some artists put fingers around microphone like a prayer. It is therefore not sufficient to point solely to the near-far-problem for justifying the margin.[28]

The frequency plans for the three cases studied are shown here (Figure 28-30):

---

[27]It can be downloaded from http://de-de.sennheiser.com/mikrofone.
[28]Eugenia Cabot of BAKOM studied this:
https://www.bakom.admin.ch/bakom/en/homepage/frequencies-and-antennas/facts-and-figures/the-effect-of-the-human-body-on-wireless-microphone-transmission.html   A study on body loss was conducted by APWPT and a paper was brought to DKE Standards organization.
https://www.apwpt.org/terminologies-a--i/body-loss-effect/index.html Their conclusion is that varying body losses also mandate 20 to 30 dB margin.
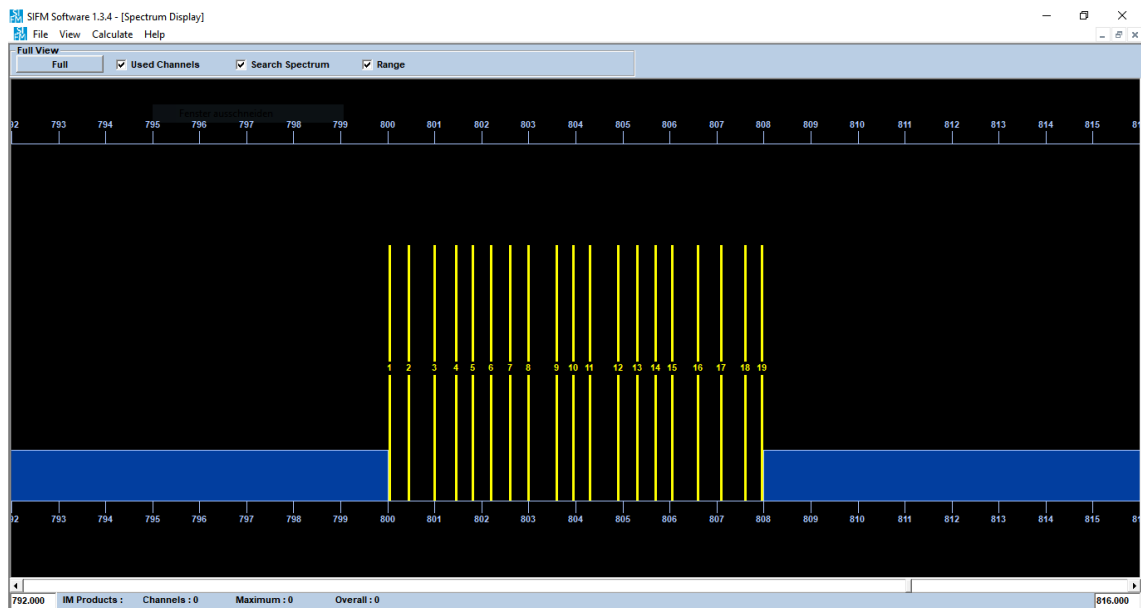
Figure 28 : Exemplary frequency plan assuming no TX intermodulation due to isolators

The frequency allocation plan in Figure 28, assuming no IM problems, does not show equal spacing. This is due to the limited tuning step size with PMSE devices. The search algorithms first places two carriers at the outer ends. Considering total 19 frequencies and subtracting two for the outer most frequencies leaves 17 frequencies to be placed in 8 MHz, which implies 18 spacings needed. Dividing 8 Mhz by 18 delivers 444.44 kHz, which cannot be selected. Either there is a 25 or 50 kHz step size typically. This implies non-equal placement of frequencies. Some spacings are a bit larger, so 450 kHz and some are a bit smaller like 425 kHz, so that on average the target spacing is met. However the exact algorithm for placing frequencies is a company secret and these algorithms are continuously further enhanced.
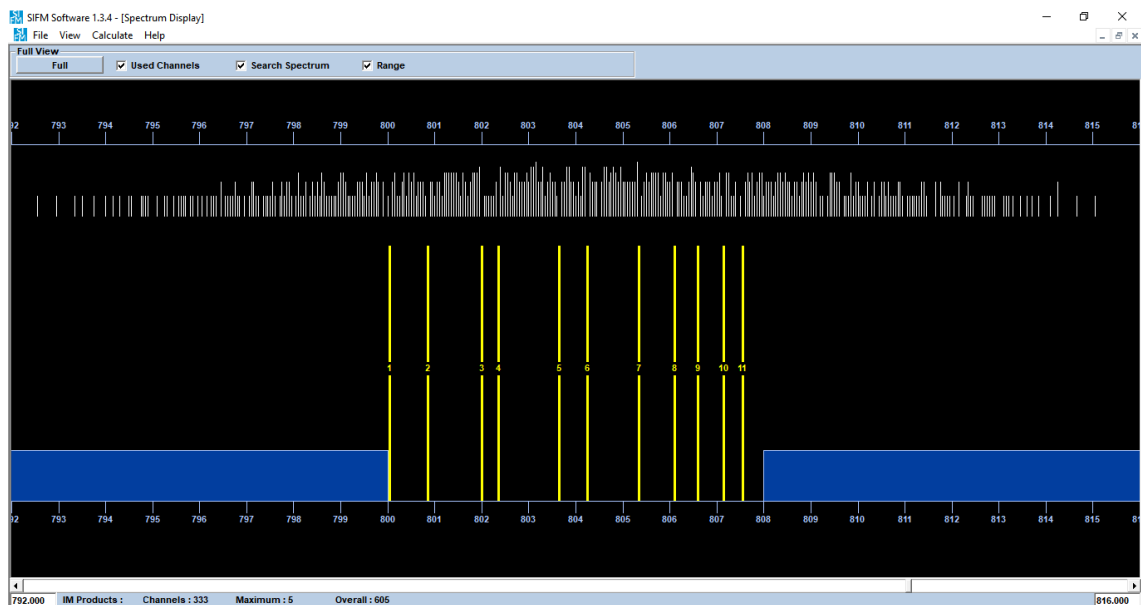

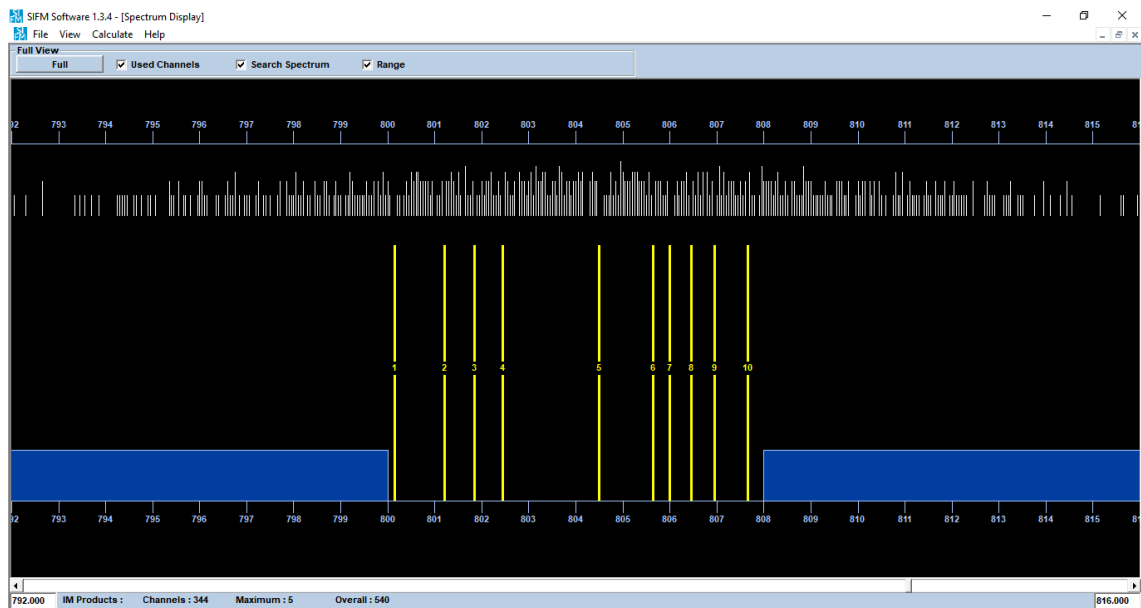**Figure 29 :** Exemplary frequency plan considering only IM3 products

Figure 30 : Exemplary frequency plan considering IM3 and IM5 products

In order not to be influenced by specific product properties and being generic, an own set for this study was defined. In essence the requirements are formulated as minimum spectral distances for IM products. The software is set to compute based on 50 kHz wide frequency slices (Figure 31).

Figure 31 : Parameter setting in SIFM for this Stratix study

Now interpreting the results from above table one can conclude the severe impact of intermodulation problems. Nearly 2x more frequencies are identified if TX intermodulation can be excluded thanks to isolators. Or in other words improved radio hardware pays off very well in terms of efficient spectrum use. Isolators allow for placing twice as many

frequencies in a TV channel. As digital PMSE systems typically have the isolator included and analogue PMSE typically not this gain is already caught when transitioning to digital.

Furthermore one can see that the impact of intermodulation gets less the higher the order. The impact by IM5 is already less, let alone IM7. That is also the justification why IM7 is not treated in the software.

If equipment allows for more dense placing of e.g. 300 kHz and frequency slices of 25 kHz are considered, above analysis changes a little bit. The following frequency plans were computed:

Table 6 : Max number of IM free frequencies in 8 MHz, assuming min spacing of 300 kHz with 25 kHz steps

	noIM	IM3	IM3_IM5	Shure Info for digital mics
Number of IM free channels	20	11	10	23

The number of 20 is lower than 8 MHz/300 kHz= 26 due to the fact that the density of packing not only is limited by IM but by the spectral shoulder, which means that shoulder power from left and right used PMSE channel add up. Now if min spacing is set to 250 kHz the following results are obtained:

Table 7 : Max number of IM free frequencies in 8 MHz, assuming min spacing of 250 kHz and 25 kHz steps

	noIM	IM3	IM3_IM5	Shure Info for digital mics
Number of IM free channels	29	12	12	23

Once again the number of 29 is lower than 8 MHz/250 kHz=32 due to the fact that power from spectral shoulders would get too high. 29 is also higher than what Shure sees as the maximum number. The issue here is that the tool provided by Sennheiser assumes discrete spectral lines at centre frequency. It is not capable of dealing with the exact spectral shaping of mic transmitters. All suppliers have to comply to spectral mask, however how close their side emission is to the spectral mask, and how it is exactly shaped is not stated. The exact spectral shaping can be influenced by transmission parameters, more precisely by the transmitter's transmission pulse shape.

When the IM problem is eliminated (case no IM) the density of packing is determined by the exact transmitters spectral shape and the margin to accommodate near-far problems. Therefore, realistic figures for the noIM case cannot be given by the tool.

From these further studies it can be concluded that closer spacing rules of 250 kHz and a fine granular choice of centre frequency in steps of 25 kHz offers a further increase in number of frequencies. However the close placing of frequencies comes at the risk of strong interference from spectral mask, which can only be managed by restricting the range an artist can move around. Dealing with the near-far problem becomes the ultimate challenge with such a frequency plan.

The fine steps of 25 kHz are also a challenge for the PLL of the transmitters. From perspective of phase noise wider steps like 50 kHz are preferred.

Aside of distance differences of microphones to receiver also fading dynamics get a problem. Deep fades are a consequence of the narrowband nature of PMSE with 200 kHz channels. A wideband system would be less prone to fading. The standards already foresee wideband PMSE systems and some PMSE vendors already have indicated working on it. The question to raise however is whether the PMSE vendors should develop their own wideband system or whether they can take over wideband technology from cellular like 5G, which is promised to meet the latency requirements from PMSE sector through URLLC (Ultra reliable low latency communication) services.

6.2   Conclusion on Scenario analysis

The density of packing PMSE channels not only is dependent on the IM performance but also by the standard, which allows for a certain spectral mask, so transmitter noise spilling of into neighbour channels.

With the use of isolators the IM problem is gone, however after eliminating the IM problem, the blocking point becomes the spectral mask. A certain level of transmitter wideband noise into neighboured spectrum is allowed by the spectral mask depicted in the EN 300 422.

It such scenarios it has to be considered that fading of wanted signal and interfering signal upper and lower of wanted channel happens independent. This gives further limits on how dense PMSE channels can be packed ensuring a minimum C/N for the wanted PMSE channel.

7 Conclusions and recommendations

The Dutch Radiocommunications Agency had two main questions for this study:

1. What type of PMSE applications that make use of the UHF band can be digitised and what types of applications cannot.
2. How much spectral savings could be obtained from digitisation of the applications that can be digitised, given the demands and constraints at large media event

The implicit assumption was that there would be strong benefits if applications could be digitised. Particularly digitisation would allow the more efficient use of spectrum, because it would be less sensitive to intermodulation. Claims of 2.5 to 3 times efficiency gain are made [8], but only under specific circumstances, in specific use cases and with high end equipment. The previous chapters show that - even when the current, more efficient, high end equipment will become more widespread - a great increase of spectrum efficiency is not achievable.

This is due to the fact that digitisation of PMSE so far only meant transition from analogue to digital transmission formats, however keeping PMSE's narrowband nature. Other systems that underwent digitisation combined the transition from analogue to digital also with a transition from narrowband to wideband air interfaces. The consequence from this is that PMSE also needs to evolve from narrowband to wideband which is already foreseen in EN 300 422 through Wideband Multichannel Audio Systems (WMAS).

For the future an increase in demand for PMSE links is predicted. Therefore not only technological assets to maintain the status quo in today's spectrum are sought, but also technological approaches to keep up with growing demand.

In narrowband systems additional unwanted intermodulation products on other radio frequencies than the original signals are generated due to intermodulation effects that happen in the RF transmit power amplifiers of PMSE equipment when an another signals enters reversely into it. The forward wanted signals intermodulate with the reversely entering signals due to nonlinearities that are always present in any analogue stage.

The immanent intermodulation problem is pronounced in the context of PMSE due to the fact that many transmitters share a physically small space and pick up each other's signals.

What the study found is that intermodulation isn't solved by digital technology per se. Intermodulation is solved by equipment that is less sensitive to intermodulation products. This is achieved by using isolators/circulators. All digital PMSE- equipment comes with isolators/circulators. Low-end analogue equipment may not have isolators/circulators on board to keep costs low. High-end analogue equipment does come with isolators/circulators and as a result do not suffer from intermodulation products either. When comparing high-end digital microphones with high-end analogue microphones it appears there is no fundamental difference in their spectrum efficiency. Any observed differences appear to be more manufacturer related, then fundamentally caused by the underlying characteristics of analogue or digital transmissions.

The strict answers to the questions of the Radiocommunications Agency are that only microphones can be digitised given the constraints of the PMSE-sector. IEMs though they could deliver digitised sound are less likely to be digitised. The sound-engineer may prefer digital microphones as they are likely to deliver a more "clean" and consistent sound. Though analogue microphones could deliver similar quality, they may not always be as clean and consistent, because of the use of companders, which have less straight forward artefacts compared to digital. It is hard to keep digital microphones under2-5ms latency. As the total latency in the chain from microphone to IEM shouldn't exceed 5ms this means that IEMs will remain analogue and the chain will be a mix of digital and analogue transmitters.

The move to higher quality microphones that use isolators/circulators is likely to allow between 16 and 23 microphones and IEMs to be planned in an 8MHz band. This will aid medium scale productions more than large scale productions. Medium scale productions will have to take IM-products less into consideration and plan around 60 devices without the aid of a dedicated spectrum planner. Larger scale events however already have a spectrum planner, they also make use of higher-end equipment, which likely already use isolators. The combination of the two means that higher-end events are already operating at around 16 (or more) devices per 8MHz and likely more if special separation etc. is available to them. If high-end equipment is used many events and locations will be able to handle even uncommon events if spectrum planning is done right. With help from the Radiocommunications Agency even exceptional events such as a crowning, a Eurovision final or a European football championship are manageable. Large scale productions will find themselves in difficulty sometimes managing all the equipment and planning it correctly, because spectrum can be limited and unforeseen other secondary use is not always easy to mitigate. This is not unmanageable; however it will require more effort and therefore investment from the event organisers.

Stratix

Literature

[1] ECC CEPT 204 Spectrum use and future requirements for PMSE Approved February 2014, http://www.erodocdb.dk/Docs/doc98/official/pdf/ECCREP204.PDF

[2] Cambridge Consultants (July 2014): Technology Evolution in the PMSE Sector, commissioned by Ofcom, P1867A-R-001 v2.1

[3] ITU-R BS.2161 (11/2009) Low delay audio coding for broadcasting applications

[4] IETF RFC1902 (1996) Structure of Management Information for Version 2 of the Simple Network Management Protocol (SNMPv2)

[5] Link M. (with Nash Technologies), Datenkommunikation im Fernbereich, Second Workshop on Intelligent Sensors: Verfahren, Protokolle und Standards zur Kommunikation medizinischer Sensor-daten, BMBF Spitzencluster Medical Valley, Erlangen, 28 June 2011

[6] Miguel Oliviera (2002), The Role of Pause Occurrence and Pause Duration in the Signaling of Narrative Structure

[7] Jin Zhou; Negar Reiskarimian; Harish Krishnaswamy (2016), Receiver with integrated magnetic-free N-path-filter-based non-reciprocal circulator and baseband self-interference cancellation for full-duplex wireless. IEEE International Solid-State Circuits Conference (ISSCC), Pages: 178 – 180

[8] EC Radio Spectrum Policy Group, DRAFT Opinion on a long-term strategy on future spectrum nee   ds and use of wireless audio and video PMSE applications, august 2017 (under consultation)

Stratix

Interviewed parties

The following people and organisations were interviewed:

Eric Pierens,	NEP Worldwide
Roland Mattijsen, Daniel Kee	Audio Electronics Mattijsen
Massimo Polo	Wisycom
Prof. Herre and Prof. Edler,	Audiolabs of FAU and Fraunhofer
Edgar Reihl, Wolfgang Bilz, Mark Brunner,	Shure
Axel Schmidt,	Sennheiser
Kees Heegstra.	Camel & co

Abbreviations

A/D	Analogue/Digital
AM	Amplitude Modulation
AT	Agentschap Telecom/Dutch Radiocommunications Agency
CBS	Dutch Central Bureau of Statistics
CEPT	Conférence européenne des administrations des postes et télécommunications
D/A	Digital/Analogue
DVB-T	Digital video broadcasting- terrestrial
FM	Frequency Modulation
IEM	In-Ear Monitor
LTE	Long Term Evolution also known as 4G mobile
MIMO	Multiple Input-Multiple Output
OFDM	Orthogonal Frequency Division Multiplexing
PMSE	Programme Making and Special Events
RA	Dutch Radiocommunications Agency
RF	Radiofrequency
RF CMOS	Radio-Frequency Complementary Metal-Oxide Semi-Conductor
SNR	Signal to Noise Ratio
TCP/IP	Transmission Control Protocol/Internet Protocol
TX	transmission power
UHF	Ultra High Frequency
VHF	Very High Frequency
VOIP	Voice over IP