Q&A: 2wcom webinar Audio over IP 1 - Basics

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List of Q&A

Question	Answers
Attendee: is the software available for download free for testing purposes?	Definitely, but you should go through a sales representative, so what we would do when you ask for a software, we would be sending you an installation document where you can download the software and install it either as a container, as your VMware image, or as even a stack of containers that provide the orchestration that you have just seen.
Berry: You can also do some testing on our live MoIN application in the cloud, is that correct?	We can also provide customized online containers. If you have a specific use case you want to do, ask a sales representative and we will set up that in the cloud. You don't even have to do all that installation stuff. We can do it for you.
Attendee: How the public IP network problem is resolving with your MoIN system?	Over the public internet it would be reliability mechanism, but it may be also a question of how you can send it over an AWS environment. In that case we can enable something like port forwarding so you can ingest streams on a specific port. And the address is not limited so you can send it to any destinations you want to do from a cloud environment. I think that is what you mean with the public IP network problem. Otherwise please specify the question again and we can jump into that.
Attendee: How synchronization can be achieved when 2 sites are on separated networks and can't share a common PTP? MPEG-TS use PCR. How is that handled in Audio over IP standards?	I would say that is a real manufacturer specific question, because the Audio of IP Sir would just say, OK, please set up a PTP network between sites and then you are able to synchronize the sites. But when you ask us or other manufacturers, then recommendation would be “use a codec from us that can supports 1PPS that is received over GPS. Or use NTP if you have Internet connection. Hence, there are different possibilities to achieve synchronization below 20 milliseconds, so that you are not able to hear the delay. But, when you use 1PPS, you can get down to microseconds again for a single frequency networks or to have a very good playout.

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Attendee: Are for SRT send and receive supported? Which types for in and out are supported to feed the SRT encapsulator? TS, ...	Yeah, that's very good. We are supporting send and receive and SRT always names them listener caller. You can decide which side should be the listener and which side should be the caller. Besides, you can send either transport stream and that encapsulates an MPTS or SPTS. Or you can also send just basic audio of IP stream. So raw audio and you can decide whether there should be an AAC codec, PCM or whatever. But when using SRT that somehow breaks interoperability because you have to select what am I receiving because SRT is not providing that information like AES67 with SDP would.
Attendee: Why is the same IP address used for different streams? There were two outputs created using the same IP multicast address for an AES67 output.	Very good. That is not the way you should use it. You should send it to different multicast addresses. Thank you for that. I did the configuration just quickly and clicked save. But it is a good thing because not every of the manufacturers can actually receive the multicar stream in that way that like we do it. We bind it also to the source. Whenever we have started to see for a multicast stream from one source, we are always receiving that stream. Even if I enable the second multicast stream, we are able to receive the first one that we have already received. Means the packets are not interfering with each other. What you recognized was a misconfiguration, but I'm happy that it was not blowing up our test set up right now.
Attendee: As capable Switches are not super expensive anymore, compared to other switches used for reliable audio production.	I would say, yes this is the case. The prices for four switches to be used and professional networks are getting lower and lower, but they're still not in a range where consumer grade switches are used, but probably don't want to use them. But I just wanted to mention that PTP could be used over them with decreased accuracy. It really depends what your network requirements are. And if you go to TV networks, it is definitely a much better accuracy that you need.
Attendee: How would I route this over WAN?	I think this refers to the virtual machine and things like that. Our software is capable to handle three different network interfaces, the control network and two data networks, so that you can just open security-based ports on one network and just data ports on the other ones. In

	addition, you could define, for example, using VMware to attach your networks, different networks that are really physically connected to your server machine. I hope that this answers the question in a way. Routing this is basically just sending multiple streams that's what I understand here. And we have also of WLAN possibilities if you need WLAN to manage your network.
Attendee: AND discovery would be limited to LAN?	**Leif:** I would say no, because it depends on how fast your multicast would be travelling, so SAP is a multicast packet that gets sent out and shows SDPs that are available on that server. if that multicast travels from your local area network to somewhere else and you can make that happen using multicasting, then also the discovery would work over your LAN. So even in a wide area network. **Hans (additional info):** SAP normally works via multicast in the LAN. With our devices, unicast could also be used to send SAP over the WAN. But currently only one SAP address is possible; no matter if multicast or unicast!
Berry: So you need multicast in that application to be able to use discovery?	**Leif:** yeah, discovery is very general. So for the session announcement protocol is for Livewire routing I am not sure I have to postpone that question if it goes for Lifewire or Dante. **Hans (additional info):** With Livewire, there is generally the possibility that other systems can connect to the Livewire routing protocol (LWRP) of a device, so that they can query the information of the device and take control. The Livewire advertisement works there together with the LWRP. This is comparable with Dante. Only Dante did not disclose this protocol in contrast to Telos. But with the IP-4c we cannot have any sources (encoder) controlled at the moment.
Attendee: What is the source IP we saw?	The source IP addresses is the IP address where the stream is coming from. Especially in case of multicast, when you have this 239.2.1.1, you cyou can not only identify the source of the stream based on the IP address

	you see. Unicast would be just be sending to a single and multicasting sending to that multicast address. The source IP address specifies where is the stream coming from. In IGMP version three, you are even able to subscribe to some specific multiclass. That is why we display it in the overview page of the web interface.
Attendee: The multicast is done by which devices? An external switch?	Multicasting in general is done using the IGMP protocol, so the device where you are running the VMware on must be capable of sending multicast. VMware is supporting that. Our software is supporting that. And then also the switch behind that must be supporting IGMP. And what we use most in our office is Cisco and Aristos switches. But those are more expensive, like Netgear manage switches, but they are also possible to be defined to send and receive forward multiple streams.
Attendee: I see buffers, how can I calculate latency caused by Moin codec?	The Buffer was just showing you the actual buffe that is used. But you can define that when you go to the elementary stream section and type in your multiclass address that you want to receive. There's also one line that shows you all your buffer and you can get that down to 15 milliseconds with MoIN so you can define your buffering and check if you're still able to receive that.
Berry: Well Leif, very good presentation. Can you do us a favor and just give us a short overview of the Q&A topics we had yesterday e.g. SIP? **Berry:** OK, so that is actually I would call it an older standard, which perfectly integrates with MoIN.	Yes, for sure, there are different things you have to consider, so SIP is definitely something where we could jump into that because that really drives interoperability. Session Announcement Protocol will not be used by the other standards. But when you do use SIP you can go over a general registry, and initiate connections between also codecs from other vendors, just not ours. So that is definitely an interesting thing where we can talk about that. Yes, that is correct.
Berry: If you want to ingest one audio your service and want to distribute it, as in multiple standards, for example, stream out in HLS, but	Yeah, that is possible, and I think that is maybe also where our software is unique because. We are able to do that at the same time, so you can take in AES67 streams and send

at the same time stream the same content to a DVB MPEG transport stream multiplexed. Is that possible? **Berry:** So, you can also do the opposite. You can ingest for example and MPEG transport stream and ingest an Icecast, take out the audio and build a new audio stream from that.	them out HLS while you send them out as DVB transport stream and at the same time you could send it out also as RTP, UDP or Icecast. Yeah, that is totally correct. We do a real transcoding. We are not doing DVB transport stream multiplexing, just taking the PID and transform it into another PID. We really decode the audio and re-encode it so you can do whatever you want with that.
Attendee: As I understood, the MoIN is a multi-source transcoder to different outputs. The question is how we will look at the hardware or will it be just software? **Berry:** Just a commercial question in between, that means if I am living in a country which has problems with importing and so on, I can buy the MoIN software from 2wcom. And for my local HP dealer, I can buy to the hardware, the server and put it on the server myself.	The MoIN is really our software solution, so you can run it on a server machine, you can run it in the cloud as well or on machines you have on premise. If you do not have hardware for that, we can suggest the hardware because it depends on the number of channels you want to run. And currently we have hardware that can run five hundred twelve channels. But for example, for a customer in the UK, we also have machines that are running four hundred fifty channels all at the same time while also running backups and all that stuff so we can take the hardware side. We are using Dell servers and HP servers for that, or you go with our fortune or codec, which is called IP-4c. So that is our real native software that can transcode as well as outputting that on XLR connectors. Yes, and do not forget about the warranty because that can also be local in your country. That can be much quicker than maybe sending it from 2wcom stock. Of course, we grant warranty, but for server PCs that are really good from Dell. We all know that.
Attendee: N/ACIP compliance is pretty important for us. Are you compliant?	Definitely, that's the EBU Tech 3326 as far as I know, and that's specifying how to do the same as AES67 using codecs so it would initiate a SIP connection. And in that connection, using the SDP would specify, OK, we are able to use these codec and we are having a complete list of codecs. We can run ACC LC LD, the high efficiency version and even the new one, this extended high efficiency or

	Opus. So, we should be pretty interoperable with a lot of vendors right now.
Berry: So, it will be a challenge to find a vendor which is not yet compliant with us.	I would say definitely one of the crucial parts is still PTP and we are also compliant with PTP using the timing. But that's more on the AES67 I think.
Berry: You mentioned IP-4c as a hardware product and MoIN as a software, if I look at the web of both products, it looks almost the same are they family members? Can you tell me something about that? It seems the software running in the IP-4c is the same as in the MoIN, or better said, is the IP-4c actually like a Dell server in a small box or how should see that?	Maybe we have to differ that a little bit because the hardware is really made by us. We design the entire hardware; we choose the processor. We do all this obsolescence management so that the hardware, when you buy an IP-4c is available for five years and more compared to a Dell server. But the software that is running inside of that is essentially the same that runs in one of these codec instances. So what we have done is we have taken our Audio over IP software, put that into a Docker container, and then wrote an orchestration so that we are able to run a lot of these codecs and just one server.
Attendee: Are there any experiences about the latency? We tried many transcoding, but the problem sometimes was the latency/delay. Is it possible to increase or decrease the buffers?	Definitely, and latency is not equal latency for some of our customers, so some are talking we need to have around 10 milli seconds and some say hundred milliseconds. Let us dive a little bit deeper into that. We can get down to 20 milliseconds of delay. And it depends on the codec as well because some codecs have a frame size, for example 24 milliseconds, then you are not able to get below that. That is one thing. And the other thing is, of course, the propagation delays. So, the real delay on your Internet connection and forward error correction would increase a lot of delay to that. I would put a lot of delay on top of that. What we suggest for Point-to-point connections would be, for example, SRT Secure Reliable Transport, because that has a fixed delay. That is always around 2.5 times the amount of time over the network. And our experience is that that is far below what forward error correction mechanisms mostly use. So I think we are somehow prepared for the latency, but feel free to ask a sales representative and connect with our engineers so that we can get a real test or whatever to make sure that we are fitting with your latency.

Attendee: Does 2wcom support FEC via SIP?	That is a good question, but more for Hans not me as Hans is our Audio over IP SIP expert. Because I am not sure if there is a command to tell that forward error correction is used in the SIP standard. But if it is possible using SIP, then I am pretty sure that we can do it because we define for our outputs the Forward Error Correction. Hans answer: Yes, we do support FEC via SIP but please have in mind also the SIP server must support FEC via SIP.
Of course, possibly only between two 2wcom devices?	That is a clarification for the SIP supporting Forward Error Correction. Seems like the gentleman already knows that it is probably not in the real standard of SIP. And then I can say, we could do something. There is a field in the SIP protocol called the agent ID, and based on that agent ID we could choose, this is a two 2wcom device and now I initiate a forward error corrected stream, for example. But since SIP calls are Point-to-point connections, I would suggest to you the SRT because SRT gives you better protection while having a lower latency. Maybe just to tell what SRT is. SRT is reliable UDP. So, it always sends packets from the sender to the receiver and only when a packet gets lost it will send another packet just so that it can still have a continuous stream. So that is why you need 2.5 times the roundtrip time to give the algorithm enough time to resend packets. But it does not have the overhead that TCP IP does because TCP IP does acknowledge every packet. And in this case, we are just sending and only when a packet gets lost. You resent that one.
Attendee: What about the ancillary data. In some solution we need to have it to sync that data with the audio. What about after the transcoding?	That really depends because ancillary data can be ingested in many ways. In a transport stream there is an ancillary data field. It can also be seen as private PID. It can be sent using the header extensions. So what we try to do is to forward that ancillary data. But we are also compliant with AES67 standard. So it depends on what codec you use and where do they put, for example, the ancillary data. But if you would use a complete 2wcom system, then on the codec side we would ingest the data as serial or UDP data (User Datagram Protocol) and we can ingest that into, for

	example, the RTP (Real Time protocol) header extensions. This also depends on the codec, because in some codecs there are ancillary data fields, and some are not. So, we have to put it in the RTP header extension. I hope that answers your question because there is a lot of things we could think of when it comes to ancillary data. Moreover, I think that transcoding question, is aiming at whether we can decode, for example, let us say somebody puts ancillary data in AES67 feed. If we decode that one and then ingest it into a real codec, for example AAC on transport stream, how is it transferred from the decoder to the encoder? And that is also something where I would have to follow up with you. It is not that easy to explain because it can be made a connection for that.
Is it possible set DSCP ToS Values rtp streams DATA1?	Yes, this is possible.