

28/2/2021

## *SIP protocol applications*

### Introduction

The EBU, foreseeing that the ISDN technology era was coming soon to its end, led the joint effort to find a non-manufacturer specific method for interchanging live audio over common IP networks. The EBU taskforce for this enterprise was the N/ACIP group. Prodys engineers joined this enterprise pretty soon and contributed with their skills; as well as took part on the series of trials among vendors (Plugtest).

As a result two specification documents were released for interchanging live audio between any two partners of different origin. Those documents are EBU TECH 3326 and its extension EBU TECH 3368.

SIP protocol reuses significantly current VoIP Internet Standards; most referred with RFC-numbers (RFC by the Internet Engineering Task Force). In order to limit variations over years and applications, the EBU TECH documents list the valid RFC references for the SIP protocol within broadcasting applications.

Although audio compression technology might evolve to achieve higher efficiency and shorter audio delay over the time, the list of solid proven compression algorithms among manufacturers is given here according the Plugtest trials and field experiences:

1. G722 (packet size 20ms)
2. MPEG L2 mono/stereo (any possible packet size)
3. PCM 16bit or Linear mono/stereo (packet size 4ms)
4. Enhanced Aptx 16bit mono/stereo (packet size 20ms)
5. MPEG4 AAC HE mono/stereo (any possible packet size)

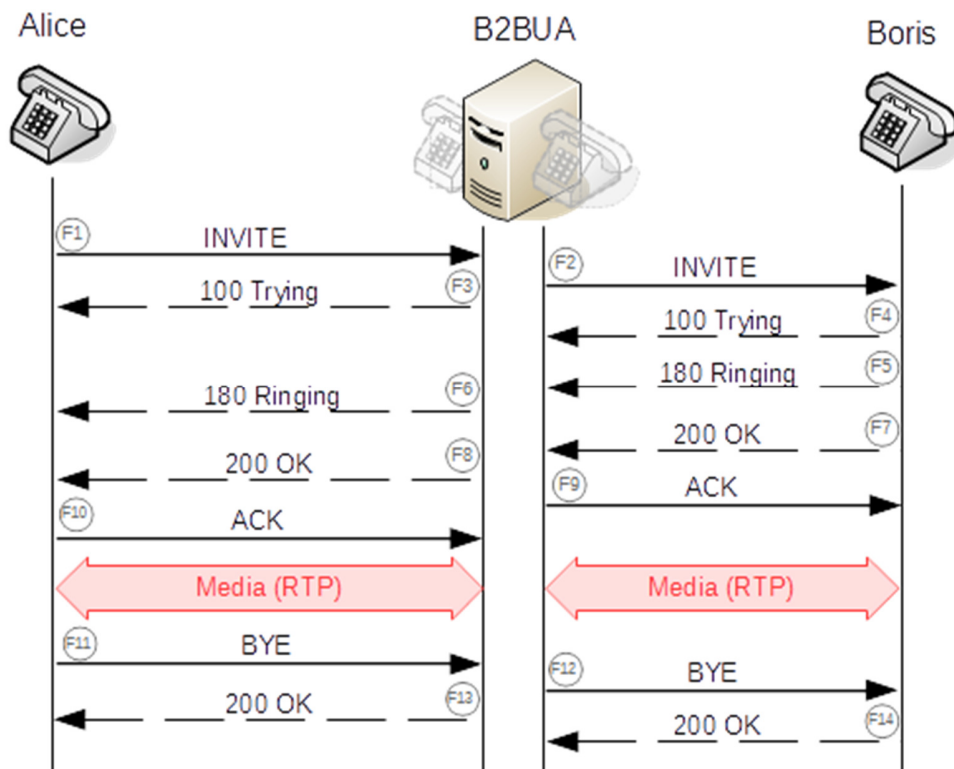


6. OPUS mono/stereo 64kbps (packet size 20ms)

Using SIP protocol, a connection starts exchanging messages over the SIP port<sup>1</sup>. The initiating end ('caller') would propose its preferred set of streaming parameters to the remote end ('callee'). The remote codec can acknowledge, propose alternative parameters or reject the invitation.

Once invite messages has been exchanged, both parties stream RTP packets out containing the audio information in a compressed format over the assigned RTP port<sup>2</sup>.

Finally any of the participants can stop the communication at its convenience.



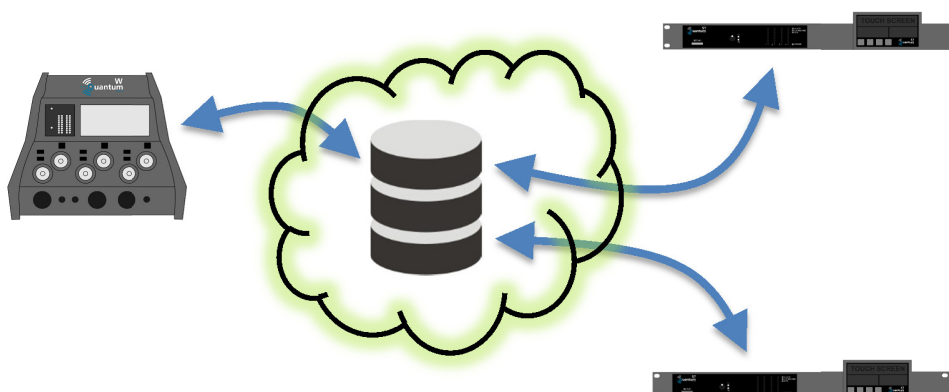
Flow diagram 1: Codec <Alice> calls codec <Boris>. Audio is finally exchanged over RTP data flow.

<sup>1</sup> Usually default SIP port is 5060.

<sup>2</sup> Usually default RTP port is 5004.

## Using SIP Server<sup>3</sup>

This is the most usual way of using SIP protocol between two codecs. The most significant fact of this arrangement is that the codecs do not interact directly, but thru a 'SIP server'<sup>4</sup> set in the middle. (please check again the icon labeled as B2BUA of the previous picture).



For this function both participants must be 'registered' on the same 'SIP server'. This statement is crucial for this application: only codecs sharing the same 'SIP Server' are enabled to communicate with each other.

SP Protocol Server Menu

☒ Register in Server

Server IP: test.sip.pr... Auth User: user3

User Name: user3 Auth Pass: \*\*\*\*\*

Timeout: 100

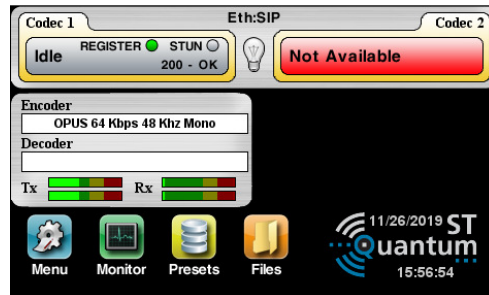
☐ Enable Outbound Proxy Server

Server:

<sup>3</sup> Please notice 'SIP Server' (sometimes referred as B2BUA SIP Server) and 'SIP Proxy Server' indicates the same concept.

<sup>4</sup> Thus for a simplified description for SIP server: this item acts similar a PABX if we would be speaking about legacy telephony/ISDN codecs.

A codec applies for its register providing login<sup>5</sup> and password. This enables further SIP operations. A 'register' remains valid while the codec remains on-line to the network or Internet. The codec takes care renewing automatically the 'register' from time to time. Prodys codecs give feedback about the current register status (**successful** [green dot]/ **unsuccessful** [red dot] or rising an alarm indication). Please refer to User Manual for details about SIP protocol parameters [IV.2.1.5]



A SIP server appears in fact to each participant to be the mate communication partner, therefore hiding location information of the codecs (e.g. actual IP address of the codecs).

Dealing with SIP server, the codec user is not required to pay attention to the real public Internet address or any other IT considerations related to forwarding ports for the local gateway.

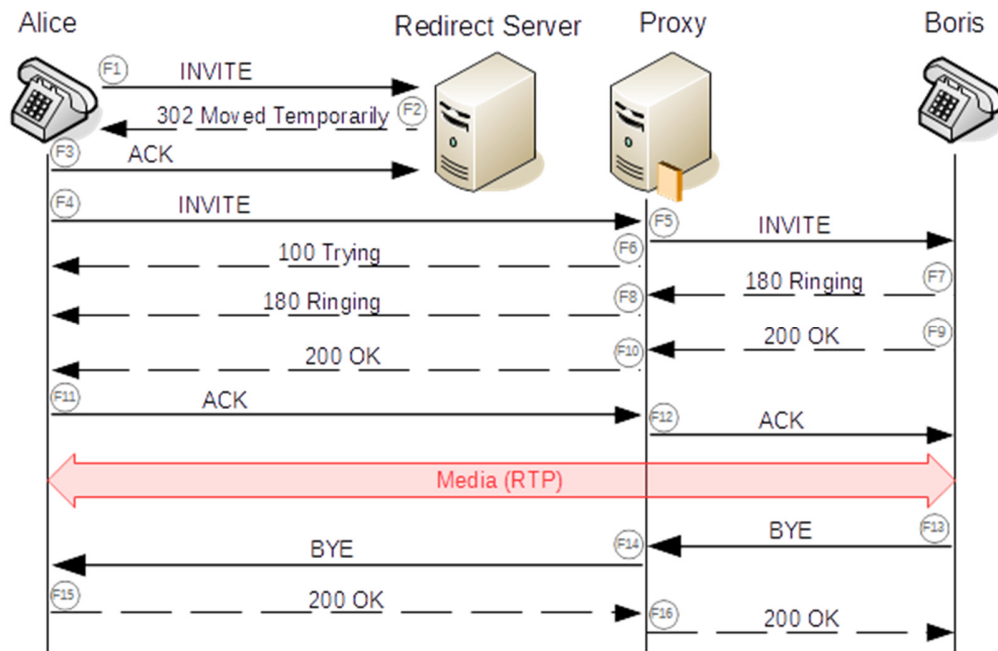
Such SIP server infrastructures support usually prevents malicious attempts over well-known ports.

## Using SIP Registrar only

Some quite simple SIP server implementations do not fulfill the B2BUA function and therefore do not pass thru the RTP stream. Those are just SIP Registrar servers. The SIP Registrar just forwards the request for initiating or stopping a communication to the other end, but both RTP streams must get their way on their own.

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<sup>5</sup> Usually the string for <Authorization User> and the <User Name> should match using Quantum codecs.

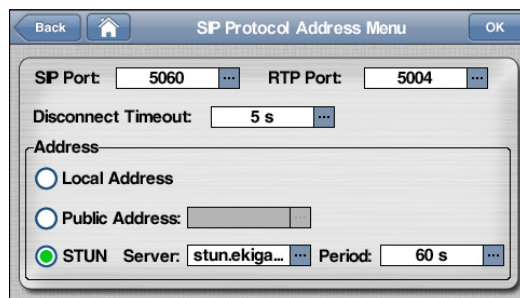


Flow diagram 2: Notice the Media dataflow RTP is directly exchanged between codecs using SIP Registrar only.

In this case the codecs must indicate their real public address before registering to the SIP registrar. This can be done either manually at codec configuration or using a complementary protocol with the name STUN.

Using SIP server without B2BUA function requires at least “open”<sup>6</sup> the RTP port 5004 on codec’s own Internet access routers and preferably enable the STUN protocol

<sup>6</sup> “open” a port thru the router means setting a PORT FORWARD definition for the codec behind.



## Using SIP protocol peer-to-peer (direct)

SIP protocol might be used for linking two codecs directly without any other SIP Server or Registrar. This method might be quite valid if two codecs from different owners must exchange audio streamed information.

Using SIP protocol directly requires “open” two ports for the communication port UDP 5060 and port UDP 5004 at the local Internet access router. In this case both codecs must provide their real public address before a connection attempt. This can be done either manually at codec configuration or using a complementary protocol with the name STUN.

It should be remarked that using this method implies some risk to experience unauthorized connection attempts<sup>7</sup>. Quantum codecs might be set for a “White List” of authorized callers up front. This function would mitigate the time of unavailability significantly. Using Quantum codecs please refer to User Manual for details about “Call Filters” [IV.2.1.16]

## Using redundant SIP servers<sup>8</sup>. SRV Look-up function.

Since the SIP server should be highly available, several methods for redundancy were worked out. One of the most powerful schema for redundancy is directly supported by Quantum codecs. This method is the SRV look-up function.

The user sets a SIP server name in the /protocol /Server menu for DNS look up. However the DNS server instead of resolving for a fully qualified IP address, replies giving a choice of possible SIP

<sup>7</sup> The main risk of those unauthorized connection attempts is they could prevent legitimated SIP users connect to the codec “under attack”.

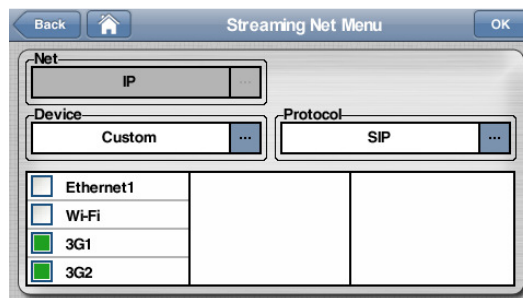
<sup>8</sup> Another simple method of using redundant SIP servers is explained at the section “*Double register on default and backup server*”

servers with some availability parameters<sup>9</sup>. In a second round the Quantum codec evaluates the availability parameters supplied and choose the most prioritized SIP server for the next operation. All this would happen at the background without further care required from the codec administrator.

## Using SIP Diversity

The SIP Diversity protocol is an extension to the regular SIP protocol as described in the EBU TECH3368 document (“Protection: Media Duplication Redundancy”). The usage of SIP Diversity is recommended for audio communication over unreliable networks.

Using SIP Diversity with a field contribution codec would enable two independent network interfaces for linking its bidirectional audio with its mate studio codec. In such way the chance of losing bits of audio information over the network would be minimized significantly.



The duplicated media stream from the field codec might “land” either on one or on two independent network interfaces of the mate studio codec, depending if a higher degree of resilience would be required for the IP connection of the codecs at the studio infrastructure.

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<sup>9</sup> For more details refer to RFC 2782.

Back		Monitor Menu				
Main	Local	Remote	Stats/sec	Total Stats		
Audio	Global	3G1	3G2			
Streaming	Lost	0	0	0		
3G/4G	Recovered	0				
System	Obsolete	0	0	0		
	Jitter	12.95	12.95	8.46		
	Tx Kbps	160	80	80		
	Rx Kbps	161	79	81		
Graph		Last Update: 18:36:37				

Back		Monitor Menu				
Main	Local	Remote	Stats/sec	Total Stats		
Audio	Global	Lan1				
Streaming1	Lost	0	0			
Streaming2	Recovered	0				
3G/4G	Obsolete	0	0			
System	Jitter	24.99	24.99			
	Tx Kbps	160	160			
	Rx Kbps	161	161			
Graph		SIP: test.sip-prodys.net:516 STUN: Disabled				

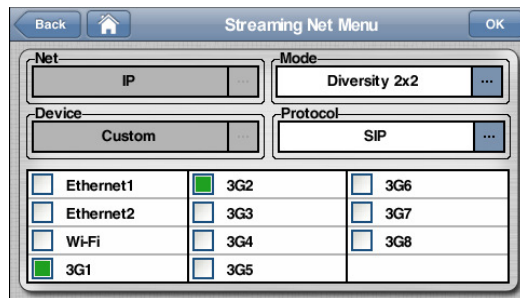
## SIP Diversity 2x2

The <SIP Diversity 2x2> protocol is an alternate way of using SIP with diversity. This method achieves higher resilience for SIP if the 3G/4G modems are in place, since one or the other cell network access may turn temporally unavailable, coming up later with its IP address changed to another number.

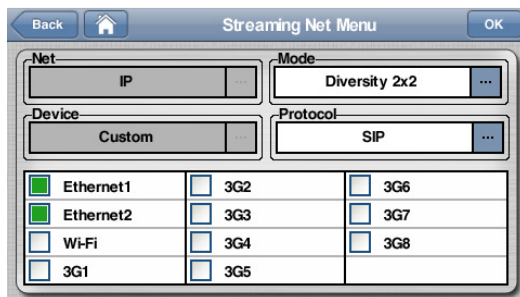
SIP Diversity 2x2 uses double SIP registration for the 'calling' and the 'receiving' codec. Therefore the customer is required to use two SIP accounts for every codec; even if one of the codecs uses just one network interface (e.g. studio codec).

Enable <SIP Diversity 2x2> protocol at the [Menu /Streaming /NET menu]:





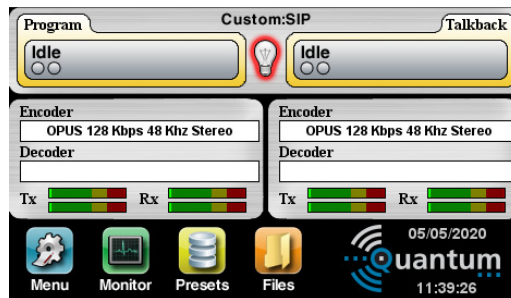
Or at the studio side:



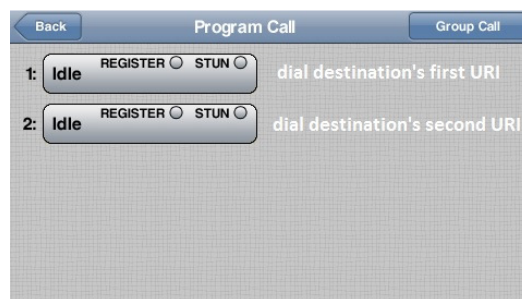
After that enter the double SIP accounts at Menu /Streaming /Protocol menu:



Upon registration, the user should verify both SIP registrations finished successfully. The connection bar should get two grey dots.



Initiating a solid SIP connection, using SIP Diversity 2x2, means dialing both destination's user URI manually of by means of a <Group Call> ( for that the BOOK should list the destination's URI).



## Double register on default and backup SIP server

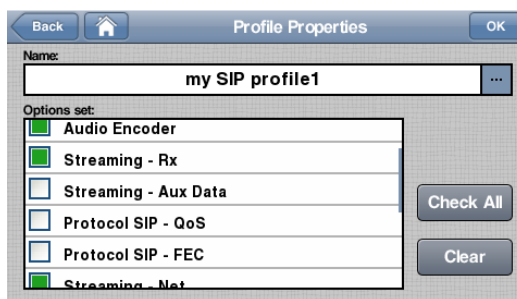
Enabling the SIP Diversity 2x2 protocol can also provide a double SIP register: the first register is aiming the 'default' SIP server, while the second register is registering on the assigned 'backup' SIP server. Actually both SIP servers, default and backup, can be on-line at the same time, but any audio connection should be routed thru only one of the possible SIP server.

## Using Profiles extension for SIP

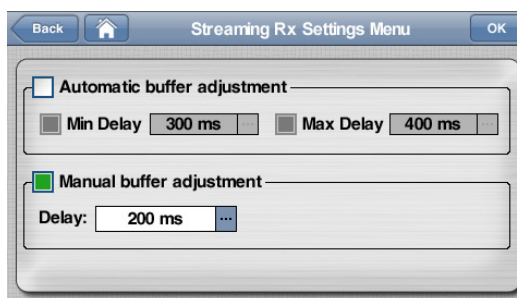
The EBU TECH3368 document describes also the usage of "Profiles" for Session Initiated Protocol (SIP). Using Profiles, the session is set up with enriched parameters for the communication. Not only the audio compression and streaming bandwidth is agreed among the communication ends, but also the optimized jitter buffer to use, QoS parameters or optimized downlink audio quality (e.g. asymmetrical audio communication).



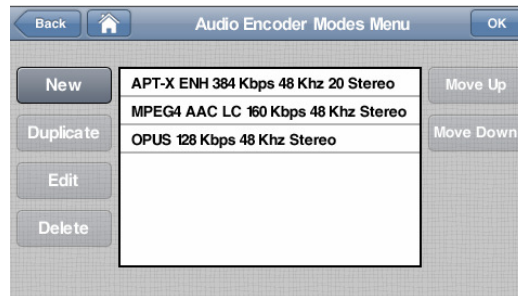
Before using profiles a Quantum owner might set up a list of useful SIP profiles at the [Menu /Streaming /Profile] menu.



The <Streaming Rx> dialog would set up the reception jitter buffer for own and far end codec involved.



The <Audio Encoder> dialog could give a priority list for the possible audio compression algorithms to negotiate with the far end codec<sup>10</sup>.



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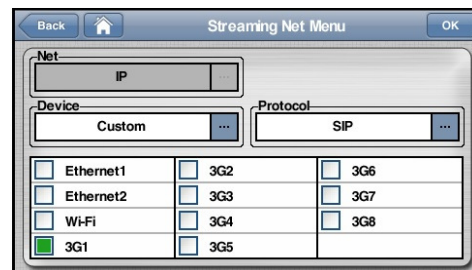
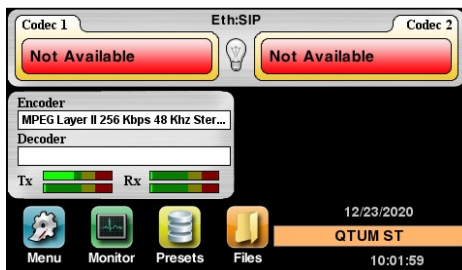
<sup>10</sup> Please notice the profile parameter <Streaming Net> should be selected for arranging using a priority list of compression algorithm. Please notice, this implies defining also the active interface or interfaces for this profile (Ethernet, 3G/4G, Custom or Wifi).

## Appendix 1:

### Troubleshooting problems using SIP protocol

This appendix tries offering some hints for solving problems if SIP protocol is in use, but seems not to work perfectly. Possible reasons for problems are listed for each case. Please verify proposed root cause and solve it.

#### Case 0: Line Not Available or Line Down

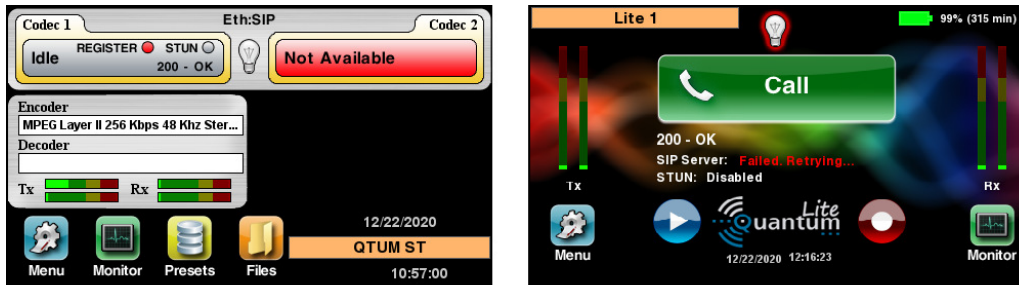


If you read on the connection bar an indication as 'Not Available' or 'Down' this means the Network Interface you attempt to use is not available to this device.

Please verify:

1. You choose at the [/Menu /Streaming /Net] the correct network interface you wish to use.
2. If you are attempting to use the Ethernet2 (rear side LAN2), you got enabled this port at the [/Menu /Interfaces /LAN /LAN2] previously.
3. If you are attempting to use any LTE modem you inserted the proper SIM cards before powering the codec ON.
4. If you are attempting to use any LTE modem of Wifi please check the User manual or tutorials about the preparation of the interface.

Case 1: SIP Register is unsuccessful since the first moment.



A red point indication means the SIP register did not end successfully. (Usually this indication remains with a yellow point indication for a short while before changing into a red point indication.) The Quantum codec will keep on trying, but probably some parameter should be corrected for getting the codec successfully registered.

Please verify:

1. You wrote the correct strings for server name and SIP URI (user login) at [**Menu /Streaming /Protocol /Codec1-or-2 /Server**]. Please notice usually the string should match for parameters 'User Name' and 'Auth User'.
2. Verify the correct 'Server IP' string or try its plain IP address.
3. Try setting the server plain IP address also as parameter 'Enable Outbound Proxy Server' [**Menu /Streaming /Protocol /Codec1-or-2 /Server**]. An example is given next.
4. Verify your interface setup got a valid DNS. For example as picture Menu /Interfaces /LAN (next picture). Please remind if you are using LTE modems the 3G menu applies; but usually this condition is perfectly met by LTE settings.
5. Is your codec granted to reach Internet using this network interface?
6. If your SIP server is using a port offset from the default 5060 port, you should specify this offset at the 'Server IP' parameter. Please check the third picture in this row. Please notice for this example the server 'sip.es' is asking for register at port :5260 instead of the default port 5060.

Back SP Protocol Server Menu OK

☒ Register in Server

Server IP: my.sip.ser... Auth User: 103491689...

User Name: 103491689... Auth Pass: \*\*\*\*\*

Timeout: 60

☒ Enable Outbound Proxy Server

Server: 193.57.201.47

Back IP Menu OK

IP Address

☒ Obtain automatically

☐ Use the following

IP: 10.10.10.215

Mask: 255.0.0.0

Gateway: 0.0.0.0

DNS

☒ Obtain automatically

☐ Use the following

Server: 10.10.2.1

Domain: prodysdom.c...

Back SP Protocol Server Menu OK

☒ Register in Server

Server IP: sip.es.5260 Auth User: user1

User Name: user1 Auth Pass: \*\*\*\*\*

Timeout: 100

☐ Enable Outbound Proxy Server

Server:

## Case 2: SIP 4xx Error message is displayed upon trying to call the mate codec



A 4xx Error message on the connection bar means the SIP Server replies on its own (e.g. 404 – Not found - online) or redirect a message from the mate codec (e.g. 488 – Not Acceptable Here). Like 4xx messages, also 5xx and 6xx Error messages might be displayed. You can find a list of standard SIP messages inside appendix 2. Please notice some vendors are using slightly different error messages for the same causes.

Please remind the last SIP Error messages on the display remains visible until a new user actions is undertaken. Therefore the original error condition could not be valid anymore.

The most common could be:

404 Not Found	: Mate codec is not registered. Message from SIP server.
488 Not Acceptable Here	: The encoder setting is not supported by the mate codec
403 Forbidden	: The encoder setting is not supported by the mate codec
486 Busy Here	: The mate is now attending currently another SIP connection
408 Requested Timeout	: No answer from mate codec or SIP server
502 Bad Gateway	: you attempted calling mate without params for SIP register

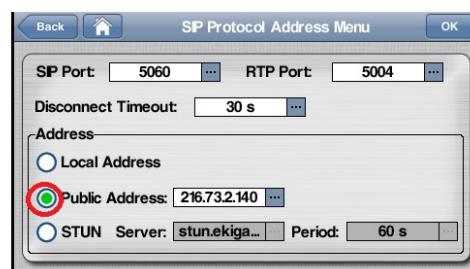
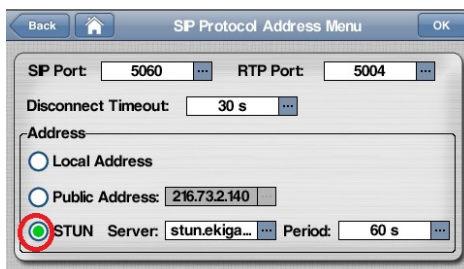


Case 3: Decoder is not playing audio out, but SIP connection was accepted for a while.

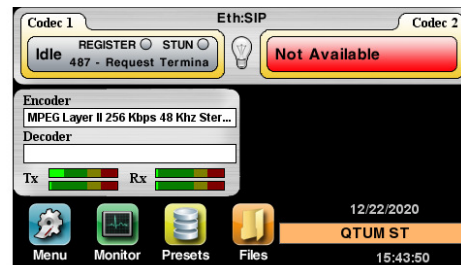
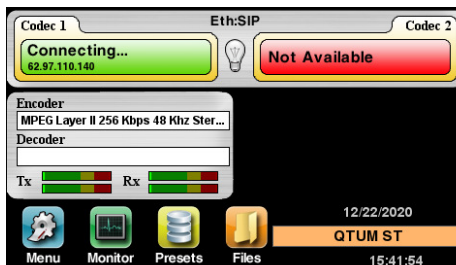


The Quantum connection bar turned green – This means the mate codec accepted the connection for streaming. If Quantum's 'Decoder' is kept 'Prebuffering' (pink) without changing into a valid decoded audio (dark blue), no audio is played out locally. Likely 30s later the connection will stop and the Quantum will be kept in idle again. Please verify:

1. The selected encoder setting are not interoperable with the mate codec. Try an MPEG L2 or and G722 encoding as an alternative to your favorite encoding.
2. If you are not using an SIP Server, but attempting a direct SIP connection (peer-to-peer), please make sure your internet router got a Port Forwarding definition for UDP Port 5004.
3. Try starting the SIP call initiating from the Quantum codec if you were contacted unsuccessfully from the mate codec.
4. If you are not using an SIP Server, but attempting a direct SIP connection (peer-to-peer), please make sure you set for a valid STUN service OR set manually your public Internet address as following example. [Menu /Streaming /Protocol /Codec1-or-2 /Address] (free STUN services are available on internet)



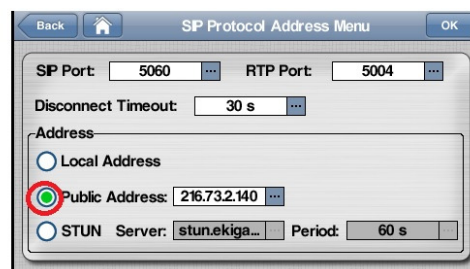
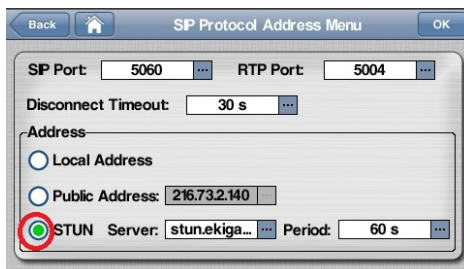
#### Case 4: Connecting – 408 Request Terminated



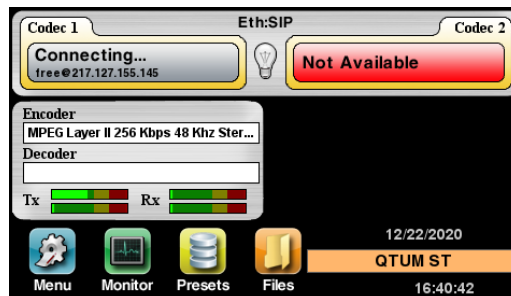
Quantum connection bar displays a blinking 'Connecting' message, but stops after some seconds with an '487 Request Terminated' Error message.

Please verify:

1. If you are not using an SIP Server, but attempting a direct SIP connection (peer-to-peer), please make sure your internet router got a Port Forwarding definition for UDP Port 5060.
2. Try starting the SIP call initiating from the mate codec if you were unsuccessfully calling from the Quantum codec.
3. If you are not using an SIP Server, but attempting a direct SIP connection (peer-to-peer), please make sure you set for a valid STUN service or set manually your public Internet address as following example. [**Menu /Streaming /Protocol /Codec1-or-2 /Address**] (free STUN services are available on internet)



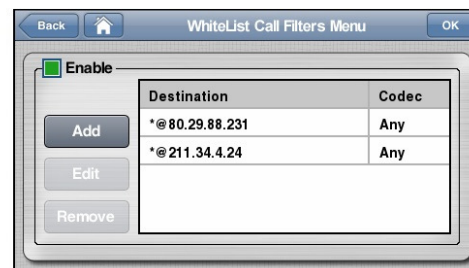
Case 5: You got many unauthorized connection attempts.



Not using SIP server for register, this is using SIP protocol for direct connection (peer-to-peer), let the codec exposed over the well-known port (5060) to unauthorized external attempts. Usually none of attempts would really get thru, but keep your codec busy frequently for several seconds.

Several countermeasures are possible:

1. Make use of a SIP server infrastructure.
2. Switch the SIP protocol on just on time to make or receive your actual SIP connection with your expected mate.
3. Enable the CallFilter function for your WhiteList [**Menu /Streaming /CallFilters /IP /WhiteList /Enable /Add**]. Define your possible mate codecs with wildcard addressing.



## Appendix 2:

### Standard SIP Messages Codes.

Code	Meaning	Code	Meaning
100	Trying	416	Unsupported URI Scheme
180	Ringing	420	Bad Extension
181	Call Being Forwarded	421	Extension Required
182	Call Queued	423	Interval Too Brief
183	Session Progress	480	Temporarily Unavailable
200	OK	481	Call/Transaction Does Not Exist
202	Accepted	482	Loop Detected
300	Multiple Choices	483	Too Many Hops
301	Moved Permanently	484	Address Incomplete
302	Moved Temporarily	485	Ambiguous
305	Use Proxy	486	Busy Here
380	Alternative Service	487	Request Terminated
400	Bad Request	488	Not Acceptable Here

401	Unauthorized	491	Request Pending
402	Payment Required	493	Undecipherable
403	Forbidden	500	Server Internal Error
404	Not Found	501	Not Implemented
405	Method Not Allowed	502	Bad Gateway
406	Not Acceptable	503	Service Unavailable
407	Proxy Authentication Required	504	Server Time-Out
408	Request Timeout	505	Version Not Supported
409	Conflict	513	Message Too Large
410	Gone	600	Busy Everywhere
411	Length Required	603	Declined
413	Request Entity Too Large	604	Does Not Exist Anywhere
414	Request URI Too Long	605	Not Acceptable
415	Unsupported Media Type		