

GETTING STARTED

PANASONIC KX-HTS SERIES

Overview

This document is a reference for configuring the Kwebbl SIP trunk into KX-NS Series systems and includes the settings required for incoming Call DDI routing and Outgoing Call CLI presentation.

Attention

This document was created based on the results of test environment accounts. We cannot guarantee SIP Trunk operation in all environments, however as a result of completing this Interoperability test we will provide technical support for any issues experienced and assist as far as possible in providing a resolution.

Codecs

- G.711 A

FAX

- T.38 not supported.

CLI

- No need for P-Asserted-Identity header and P-Preferred-Identity header.

CLIR Calls Limitation

- HTS don't support CLIR Outgoing Calls (Anonymous)

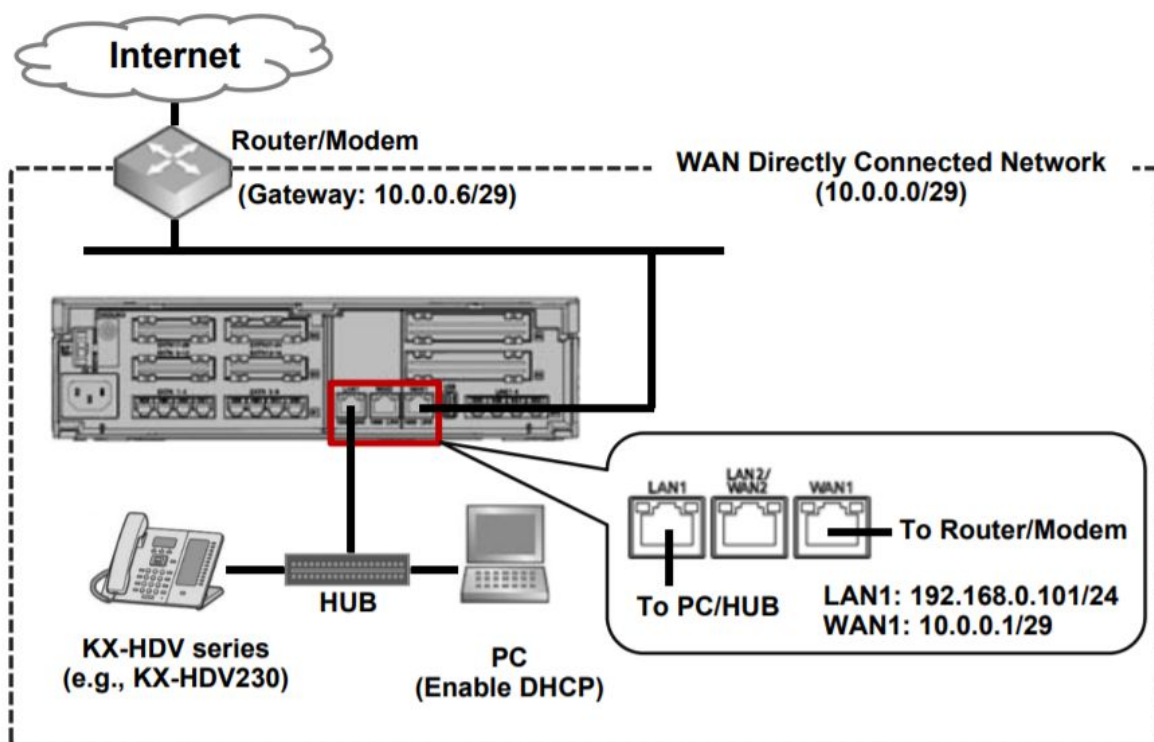
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External Router Example

HTS has a Built-in Router functions so advanced routing configuration can be managed by the HTS and the internet router/modem simply providing internet access.

WAN Network: 10.0.0.0
Subnet Mask: 255.255.255.48 /29
PBX assigned WAN-1 IP: 10.0.0.1
WAN Gateway IP address: 10.0.0.6
Static DNS
Primary DNS Server: 8.8.8.8
Secondary DNS Server: 8.8.4.4



Note

Configure the settings according to the IP address system of the mapped WAN. Also, set the IP address which does not duplicate other IP devices. .

1. SIP Trunk

SIP Extension & SIP Trunk Common

Select [PBX Configuration] - [2. Extension] - [7.SIP Extension Property]

SIP Extension & SIP Trunk Common		
Voice (RTP) UDP Port No. (Server)	12000	(1024 - 57435)
SIP Port Number	5060	(1024 - 65535)
Jitter Buffer Type for Voice	Adaptive	▼
Jitter Buffer Delay Min. for Voice	20	(ms) ▼
Jitter Buffer Delay Max. for Voice	180	(ms) ▼
Jitter Buffer Delay Init. for Voice	50	(ms) ▼
Jitter Buffer Type for Data	Fixed	▼
Jitter Buffer Delay Min. for Data	20	(ms) ▼
Jitter Buffer Delay Max. for Data	180	(ms) ▼
Jitter Buffer Delay Init. for Data	50	(ms) ▼
NAT Traversal	<input checked="" type="radio"/> Off <input type="radio"/> Fixed IP Address <input type="radio"/> STUN <input type="radio"/> DDNS	
NAT - Fixed Global IP Address	0 . 0 . 0 . 0	
NAT - STUN Server	0.0.0.0 [0-9 a-z A-Z - .]	
NAT - Host Name	0.0.0.0 [0-9 a-z A-Z - .]	
NAT - STUN/DDNS Refresh Interval	300	(s) ▼
RTP Inactive Timer	0	(s) ▼

Apply Cancel

1. Voice(RTP) UDP port No.(Server): Leave at 12000-
2. SIP Port Number: Leave at 5060
3. NAT Traversal: Leave Off
4. NAT – Fixed Global IP Address: Leave at 0.0.0.0

All other settings can be left at default.

Click [Apply]

Click [Apply Now], If you want to apply the settings immediately. Active calls will be disconnected.
Click [Apply Later], If you want to apply the settings when there is no active call.
Click [Cancel], Close this dialog without apply the settings.

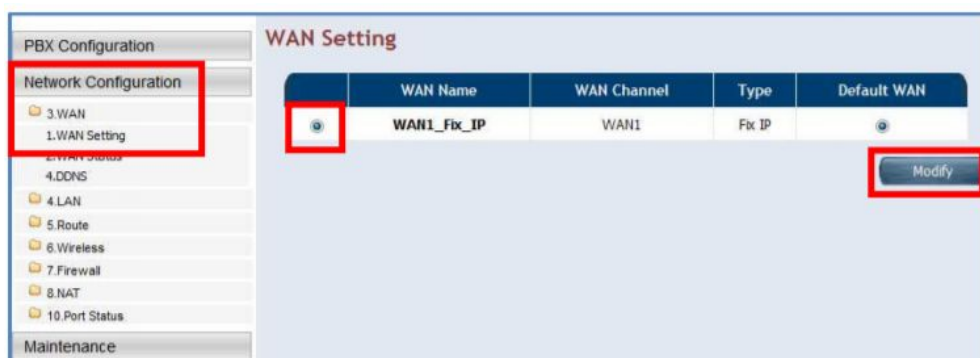
Apply Now Apply Later Cancel

Click [Apply Now]

2. Network Settings

Network Configuration WAN Setting

Select [Network Configuration] - [3.WAN] – [1.WAN Setting] and Select [WAN 1 Channel] Click [Modify]



HTS WAN-1 IP: 10.0.0.1 (Example)

Select the appropriate the Connection Type. In this example, use the Static IP address Option.

HTS can be connected to your service provider in any of the following ways

WAN TYPE: Fix IP

WAN connection Name: WAN1_Fix_IP

IP address, by your ISP: 10 . 0 . 0 . 1

Subnet Mask: 255 . 255 . 255 . 248

ISP Gateway Address: 10 . 0 . 0 . 6

Static DNS:

Primary DNS Server: 8 . 8 . 8 . 8

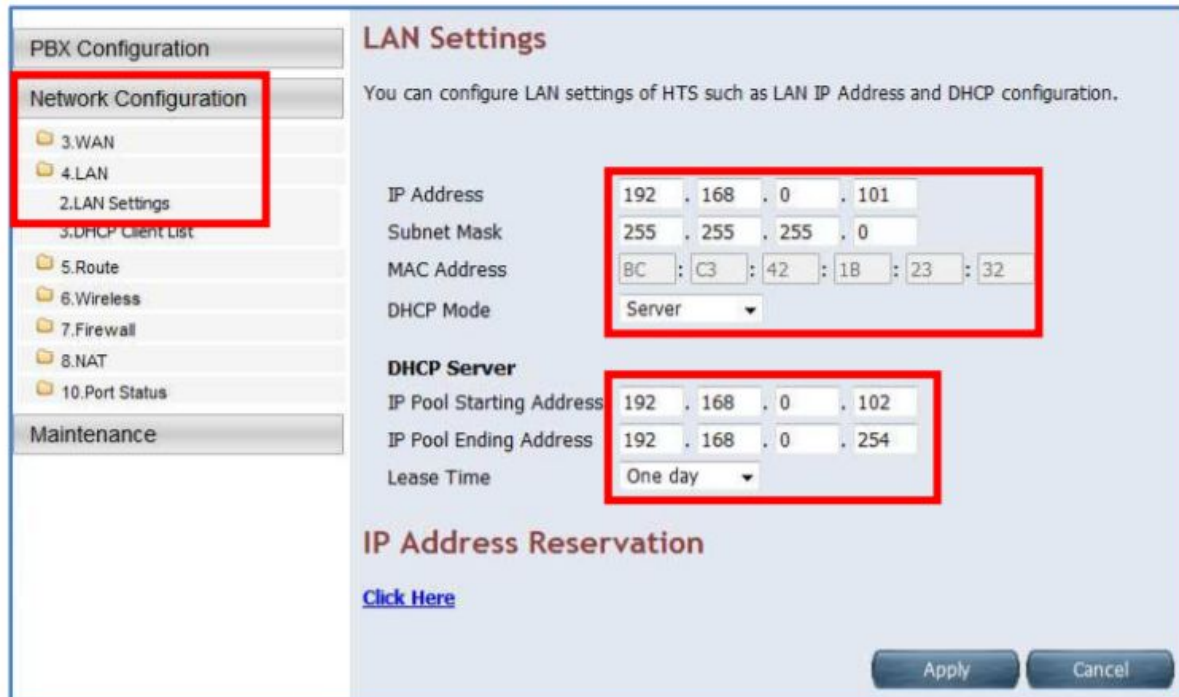
Secondary DNS Server: 8 . 8 . 4 . 4

1. WAN Type: Fix IP
2. WAN Connection Name: Leave WAN1_Fix_IP - Reference only
3. IP address, by your ISP: (Enter your actual IP address)
4. Subnet Mask: (Enter your appropriate subnet mask)
5. ISP Gateway Address: (Enter the actual Gateway IP address)
6. Static DNS: Change to Enable
7. Primary DNS Server: (Enter the actual DNS IP address)
8. Secondary DNS Server: (Enter the actual Secondary DNS IP address)

Click [Apply]

LAN Settings

Go to [Network Configuration] - [4.LAN] - [2.LAN Settings]



- 1. IP Address: 192.168.0.101
- 2. Subnet Mask: 255.255.255.0
- 3. DHCP Mode: Server

[DHCP server]

- 4. IP Pool Starting Address: Leave 192.168.0.102
- 5. IP Pool Ending Address: Leave 192.168.0.254
- 6. Lease Time: Leave One day

3. Trunk Port Attribution

Select [PBX Configuration] - [3.Trunk] - [1.Port] and click [Edit / CO Line Number 1 to 8]



[CO Line Number 1]

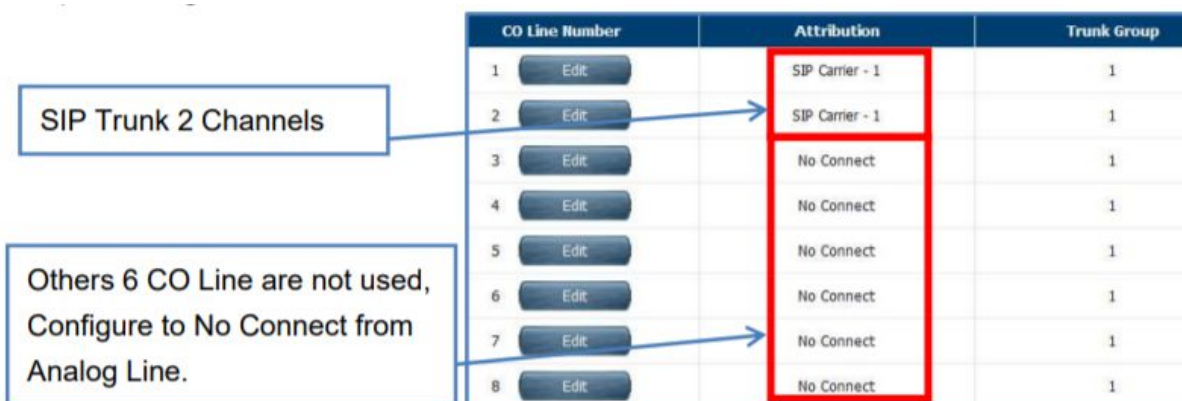
- 1. Attribution: SIP Carrier - 1
- 2. Trunk Group: Leave at 1

Click [Apply]

Click [Apply Now]



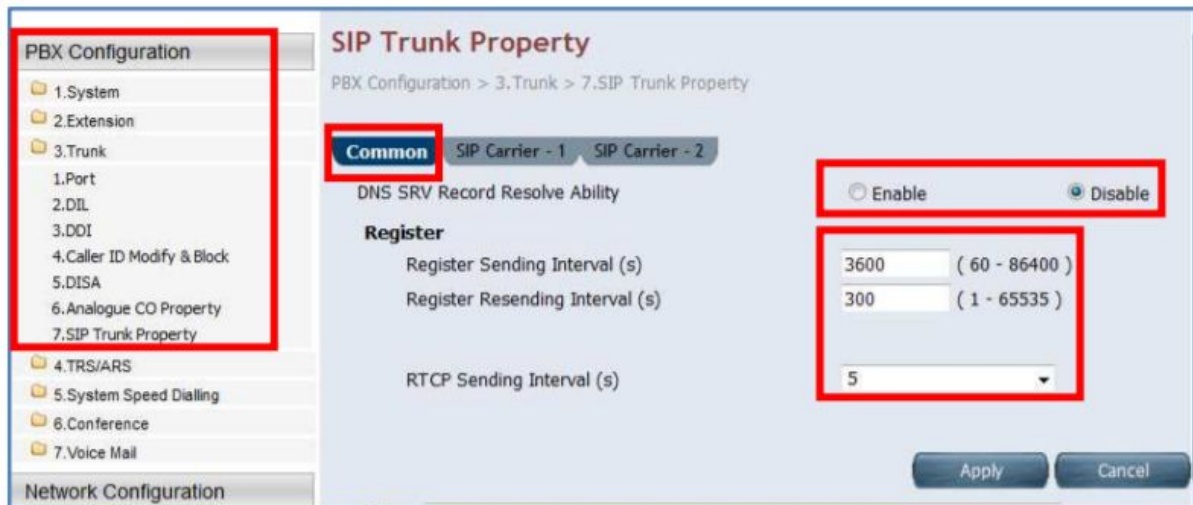
Example, Configure "CO-Line 1-2: SIP Carrier-1, Other CO-Line: No Connect"



4. Configuring a SIP Trunk

SIP Trunk Property - Common

Select [PBX Configuration] - [3.Trunk] - [7. SIP Trunk Property] - [Common]



1. DNS SRV Record Resolve Ability: Leave Disable

[Register]

2. Register Sending Interval (s): Leave at 3600

3. Register Resending Interval (s): Leave at 300

4. RTCP Sending Interval (s): Leave at 5

SIP Carrier-1

- | | |
|----------------------------|-------------------------------|
| 1. Provider Name: | Enter a name – reference only |
| 2. SIP Server Name: | 1.trunk.sip.kwebbl.net |
| 3. SIP Server IP Address: | Leave at 0.0.0.0 |
| 4. SIP Server port Number: | Leave at 5060 |
| 5. SIP Server Domain: | Not required |
| 6. Subscriber Number: | Not required |

[Account]

- | | |
|-----------------------------|--|
| 7. User name: | Enter the SIP Account (User name) as supplied by Kwebbl.
(Note this is user name without @sip.eqinox.com)
For example: SIP Account (User name) = account1 Enter: account1 |
| 8. Authentication ID: | Enter the Authentication ID as supplied by Kwebbl.
(Note this is user name without @sip.eqinox.com)
For example: Authentication ID = account1 Enter: account1 |
| 9. Authentication Password: | Enter the Authentication Password as supplied by Kwebbl.
For example: Authentication Password = passABCD Enter: passABCD |

[Register]

- | | |
|-----------------------|--------|
| 10. Register Ability: | Enable |
|-----------------------|--------|

Calling Party

Header Type From Header
 P-Preferred-Identity Header
 P-Asserted-Identity Header

Host Name [0-9 a-z A-Z - .]

Format Without "+" With "+"

Send CLIP of caller when call from SIP carrier is forwarded to same SIP carrier Enable Disable

Called Party

Format Without "+" With "+"

Header Type (Receive) Auto
 Request-URI
 To Header
 P-Called-Party-ID Header

Number Check Ability Enable Disable

[Calling Party]

1. Header Type: Leave From Header
2. Host Name: Not required
3. Format: Leave Without "+"
4. Send CLIP of caller when call from SIP Carrier is forwarded to same SIP carrier: Leave Enable

[Called Party]

5. Format: Leave Without "+"
6. Header Type (Receive): Leave Auto
7. Number Check Ability: Leave Disable

Voice

IP Codec Priority 1st: G.711A
 G.711Mu
 G.729A
 G.711A

IP Codec Priority 2nd: G.711Mu
 G.729A
 None
 G.711A
 G.711Mu
 G.729A

IP Codec Priority 3rd: None

Packet Sampling Time: 20 (ms)

DTMF: Inband Outband(RFC2833)

Payload Type (DTMF): 101

Session

Session Timer Ability: Enable(Passive) Enable(Active) Disable

Session Expire Timer (s): 180 (90 - 3600)

Session Incoming Refresher Request: UAC UAS

Session Refresh Method: re-INVITE UPDATE

[Voice]

1. IP Codec Priority 1st: Leave G.711A
2. IP Codec Priority 2nd: Leave G.711Mu
3. IP Codec Priority 3rd: Leave None
4. Packet Sampling Time: Leave at 20 (ms)
5. DTMF: Leave Outband (RFC2833)
6. Payload Type (DTMF): Leave at 101

[Session]

7. Session Timer Ability: Leave Enabled (Passive)
8. Session Expire Timer: Leave at 180
9. Session Incoming Refresher Request: Leave UAC
10. Session Refresh Method: Leave re-INVITE

Click **[Apply]**

Click **[Apply Now]**

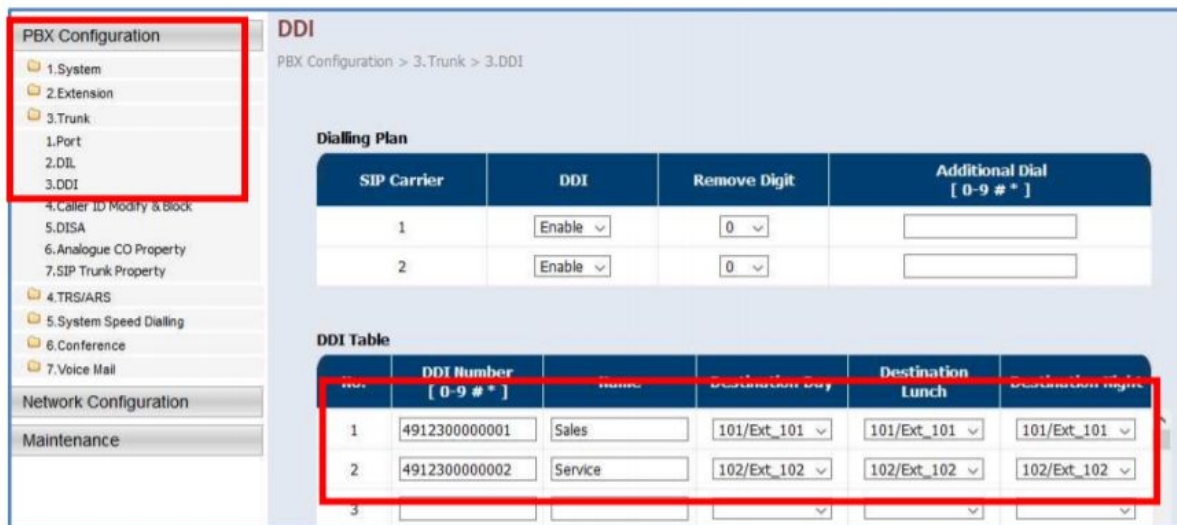
Click **[Save & Logout]**

HTS Web Maintenance Console 001.91011 English (UK) **[Save & Logout]**

5. Incoming Call Routing

DDI Table

[PBX Configuration] - [3.Trunk] - [3.DDI]



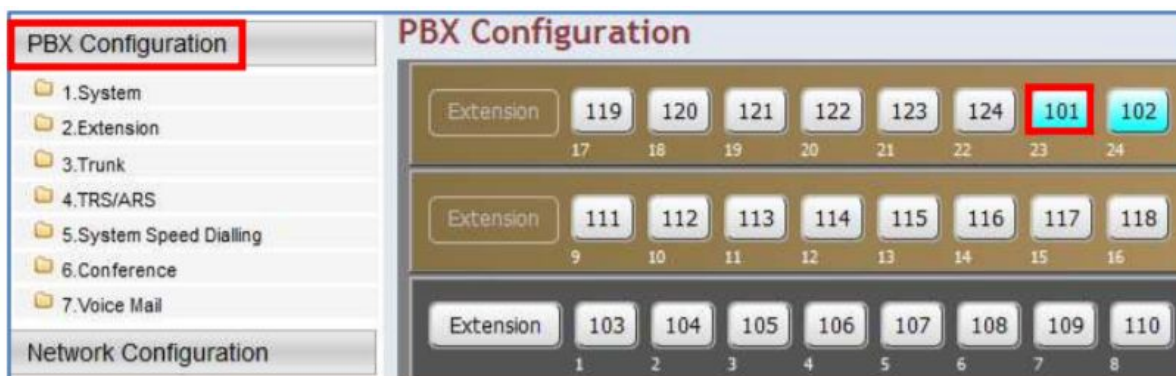
- 1. DDI Number: Enter the DDI number in the format (as below)
Example Number: 49 123 0000001 (DDI Number) Enter: 491230000001
- 2. Name: Determined by the installer (optional setting)
- 3. Destinations: Determined by the installer (extension number, group etc)

Click **[Apply]**

6. Outgoing Call CLI

CLIP

[PBX Configuration] - [Target Extension] Click (Example: Extension 101)



[Port - Extension 101 / CLIP]

Port
PBX Configuration > 2.Extension > 1.Port

Extension Number	101 [0-9]
Extension Name	HTS Ext Name101
Attribution	SIP 23
FAX Connection	<input type="radio"/> Yes <input checked="" type="radio"/> No
Manager	<input checked="" type="radio"/> Enable <input type="radio"/> Disable

CLIP

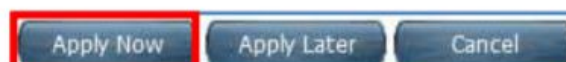
SIP Carrier - 1	4912300000001 [0-9]
SIP Carrier - 2	[0-9]

1. Extension Name: HTS Ext Name101 (Example)
2. SIP Carrier -1: 4912300000001 (Example)

Click [Apply]



Click [Apply Now]



Click [Save & Logout]



7. Appendix

UDP hole punching for keeping in Router/Firewall Port Forwarding

If Kwebbl's Keep Alive message (OPTIONS UDP packets) is not effect to keep port forwarding for external router, configure the OPTIONS packet to enable as keep-alive message on the PBX SIP Trunk Property.

The screenshot shows the PBX Configuration interface. On the left, there is a navigation tree with categories: PBX Configuration, Network Configuration, and Maintenance. Under PBX Configuration, items include 1.System, 2.Extension, 3.Trunk, 4.TRS/ARS, 5.System Speed Dialling, 6.Conference, and 7.Voice Mail. The '3.Trunk' category is expanded, showing sub-items: 1.Port, 2.DIL, 3.DDI, 4.Caller ID Modify & Block, 5.DISAS, 6.Analogue CO Property, and 7.SIP Trunk Property. The '7.SIP Trunk Property' item is selected, and its configuration is displayed in the main area.

The configuration is divided into three sections:

- Session:**
 - Session Timer Ability: Enable(Passive) Enable(Active)
 - Session Expire Timer (s): 180 (90 - 3600)
 - Session Incoming Refresher Request: UAC UAS
- DSP:**
 - IP Side - Gain1 (Network to PBX): +0 (dB)
 - IP Side - Gain2 (PBX to Network): +0 (dB)
 - PCM Side - Gain1 (PBX to Network): +0 (dB)
 - PCM Side - Gain2 (Network to PBX): +0 (dB)
 - PCM Side - Echo Canceller Type: NE (dB)
 - PCM Side - Echo Canceller NLP: On
 - PCM Side - Echo Canceller Window Size: 6 (ms)
- Option:**
 - ITSP Port Check: Enable Disable
 - Alive Check: Enable Disable
 - Alive Check - Interval: 60 (seconds)

If necessary, For External router setup, configure Port Forwarding on the router as follows: udp port 5060 – to HTS WAN1 IP address (e.g. 192.168.22.99)

udp port range 12000-20000 (RTP) – to WAN1 IP address (e.g. 192.168.22.99)

IMPORTANT!

To secure the PBX from illegal attacks, please restrict the above port forwarding ports to only be accessible from the Kwebbl source IP addresses.