TB Academy

*T*media™ Gateways – Essentials and Configuration

eLearning Activity Book (with answers)

March 2017

Release: 2.9.65

UNIT 3

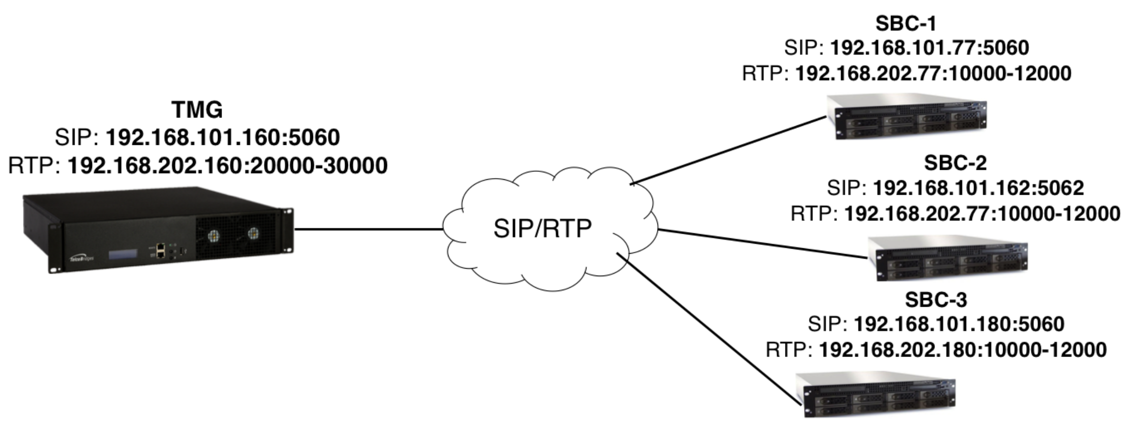
SIP

3.1 SIP Configuration

3.1.1 Activity

⌚ 30 minutes

In this activity, you will configure SBC-1, SBC-2 and SBC-3 SIP peers.



Local SIP/RTP

- SIP: 192.168.101.160, UDP port 5060
- RTP: 192.168.202.160, UDP ports 20000-30000

Remote SIP/RTP

1. SIP IP ToS value: 32 (decimal)
2. “SBC-1” SIP peer
   - SIP: 192.168.101.77, port 5060, UDP
   - RTP: 192.168.202.77, UDP ports 10000-12000
3. “SBC-2” SIP peer
   - SIP: 192.168.101.162, port 5062, UDP
   - RTP: 192.168.202.162, UDP ports 10000-12000
4. “SBC-3” SIP peer
   - SIP: 192.168.101.180, port 5060, UDP
   - RTP: 192.168.202.180, UDP ports 10000-12000

✍ **SIP Type of Service (ToS)** is now configurable with TMG-CONTROL 2.9

For the system connection information, refer to the file ***"TMGEC_lab_connection_info.pdf"*** :

1. Open your web browser, type the http address of the lab system
2. Login with the given ***"Username"*** and ***"Password"***
3. Select your configuration

From the web portal, execute the following steps.

✔ Step 1 - Create the SIP Configuration

✍ Each Hardware Unit needs its own SIP Stack to support SIP trunks

1. Click on ***"SIP"*** menu
2. Click on ***"Create New SIP"***
   a) In ***"Name"***, type ***"sip_UNIT01"***
   b) In ***"Unit"***, select UNIT01
   c) Click on ***"Create"***

✔ Step 2 - Create the SIP Transport Server

✍ The same ***"Transport Server"*** is used to carry SIP for "SBC-1", "SBC-2" and "SBC-3".

1. Click on ***"Create New Transport Server"***
   a) In ***"Name"***, type ***"TS_101_160_5060"***
   b) In ***"Port Type"***, select ***"UDP"***
   c) In ***"Port"***, type ***"5060"***
   d) In ***"IP Interface"***, select ***"sip_101_160"***
   e) Click on ***"Advanced"***
   f) In ***"IP Header Type of Service (ToS)"***, type ***"32"***
      ✍ See Appendix section A.1 on page 8 for documentation on QoS
   g) Click on ***"Create"***

✔ Step 3 - Create a "SBC-1" SIP NAP

1. Click on ***"NAPs"*** menu
2. Click on ***"Create New NAP"***
   a) In ***"Name"***, type ***"SBC_1"***
   b) Click on ***"Create"***
   c) In ***"SIP Transport Server"***, add ***"TS_101_160_5060"***, wait for the page to reload
   d) In ***"Proxy address"***, type ***"192.168.101.77"***
   e) In ***"Proxy port type"***, select ***"UDP"***
   f) In ***"Proxy port"***, type ***"5060"***
   g) In ***"Port range"***, add ***"pr_rtp_202_160"***
   h) Click ***"Save"***

✔ **Step 4 -** Create a "SBC-2" SIP NAP

1. Click on ***"NAPs"*** menu
2. Click on ***"Create New NAP"***
   a) In ***"Name"***, type ***"SBC_2"***
   b) Click on ***"Create"***
   c) In ***"SIP Transport Server"***, add ***"TS_101_160_5060"***, wait for the page to reload
   d) In ***"Proxy address"***, type ***"192.168.101.162"***
   e) In ***"Proxy port type"***, select ***"UDP"***
   f) In ***"Proxy port"***, type ***"5062"***
   g) In ***"Port range"***, add ***"pr_rtp_202_160"***
   h) Click ***"Save"***

✔ **Step 5 -** Create a "SBC-3" SIP NAP

1. Click on ***"NAPs"*** menu
2. Click on ***"Create New NAP"***
   a) In ***"Name"***, type ***"SBC_3"***
   b) Click on ***"Create"***
   c) In ***"SIP Transport Server"***, add ***"TS_101_160_5060"***, wait for the page to reload
   d) In ***"Proxy address"***, type ***"192.168.101.180"***
   e) In ***"Proxy port type"***, select ***"UDP"***
   f) In ***"Proxy port"***, type ***"5060"***
   g) In ***"Port range"***, add ***"pr_rtp_202_160"***
   h) Click ***"Save"***

✎ Answer the Questions

1. Do you need to configure the remote RTP IP and port range? _______*[A1] [No]*_______
2. Why? _______*[A2] [this is configured on the peer and provided by the peer in the SDP of the SIP messages]*_______

☛ Validate your Results

- Click on ***"SIP"*** menu, and then click on ***"sip_UNIT01"*** to verify:

Transport Servers

Name	Port	Port Type	IP interfaces
TS_101_160_5060	5060	UDP	sip_101_160

- Click on ***"NAPs"*** menu to verify:

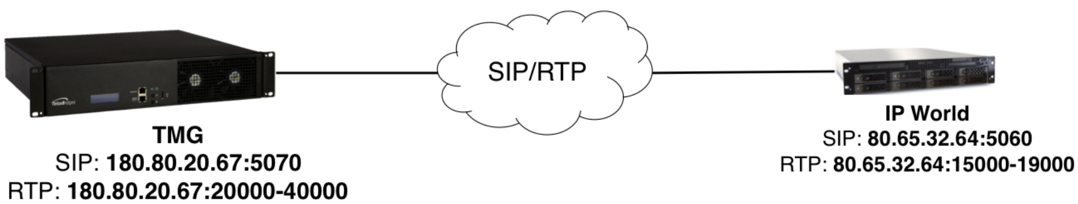
Network Access Point List

Name	Profile	Channel Usage	SIP Proxy	Members
SBC_1	default		UDP 192.168.101.77:5060	TS_101_160_5060, UNIT01:sip_101_160:pr_rtp_202_160
SBC_2	default		UDP 192.168.101.162:5062	TS_101_160_5060, UNIT01:sip_101_160:pr_rtp_202_160
SBC_3	default		UDP 192.168.101.180:5060	TS_101_160_5060, UNIT01:sip_101_160:pr_rtp_202_160

3.1.2 Exercise *(Optional)*

◷ 10 minutes

In this exercise, you must configure IP World SIP peer with the following information:



Local SIP/RTP

- SIP: 180.80.20.67, UDP port 5070
- RTP: 180.80.20.67, UDP ports 20000-40000

Remote "IP World" SIP/RTP

- SIP: 80.65.32.64, port 5060, UDP
- RTP: 80.65.32.64, UDP ports 15000-19000

✍ You must have completed the optional exercise of Section ***"IP configuration"*** in Lesson ***"IP Network"***

✎ Answer the Questions

1. How many new SIP Stacks will you configure? ______*[A3] [0]*______
   Why? ______*[A4] [configure only one SIP stack per telecom unit]*______
2. How many new SIP Transport Servers will you configure? ______*[A5] [1; 180.80.20.67:5070]*______
3. How many NAPs must be created? ______*[A6] [1; 80.65.32.64:5060]*______

➤ Start the Exercise

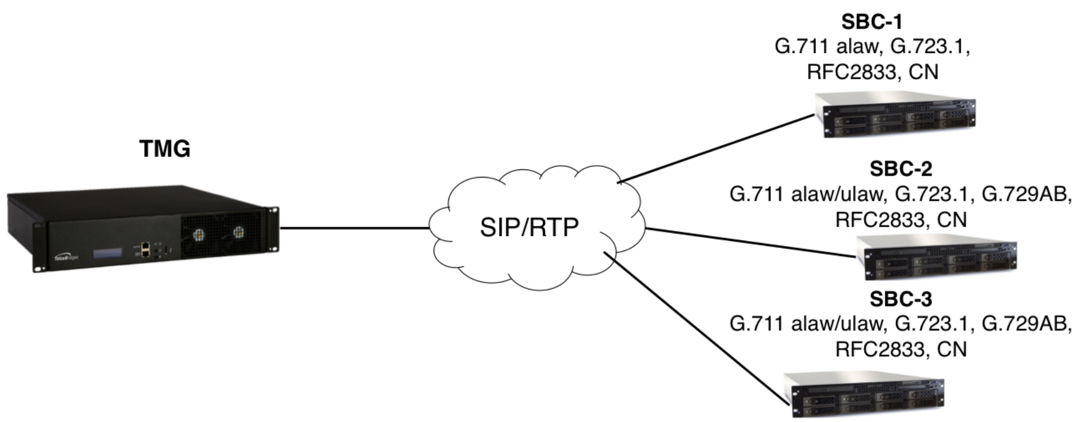
1. Select your configuration
2. Configure the required SIP transport server(s) and SIP NAP(s)
3. Validate your configuration from Appendix subsection B.1.1 on page 9

3.2 SIP Codecs Configuration

3.2.1 Activity

10 minutes

In this activity, you will change SBC-1 VoIP codecs and RTP IP ToS value



RTP ToS value for all SBCs: 32 (decimal)

SBC-1 codecs:

- In order or priority: G.711 aLaw and G.723.1
- Relay events 0 to 15 and 32 to 36 (RFC2833) with Payload Type 101
- Voice activity detection (Confort Noise) with Payload Type 13

SBC-2 and SBC-3 codecs (default profile):

- In order or priority: G.711 uLaw, G.722 aLaw, G.723.1 and G.729AB
- Relay events 0 to 15 and 32 to 36 (RFC2833) with Payload Type 101
- Voice activity detection (Confort Noise) with Payload Type 13

For the system connection information, refer to the file ***"TMGEC_lab_connection_info.pdf"*** :

1. Open your web browser, type the http address of the lab system
2. Login with the given ***"Username"*** and ***"Password"***
3. Select your configuration

From the web portal, execute the following steps.

✔ **Step 1 -** Change RTP ToS in default profile

1. Click on ***"Profiles"*** menu
2. Click on ***"default"***

    a) Click on ***"RTP and Audio"***

    b) In ***"Packet Network"***,

i. In ***"Type of Service (ToS)"***, type ***"32"***
✍ See Appendix section A.1 on page 8 for documentation on QoS

c) Click on ***"Save"***

✔ **Step 2 -** Create a new Profile for "SBC-1"

1. Click on ***"Profiles"*** menu
2. Click on ***"Copy"*** of the ***"default"*** profile

a) In Name, type ***"default_g711_g723"***

b) Click on ***"Copy"***

i. Why do we copy the profile instead of modifying it directly?______________________

______*[A7] [specific profile that will be used only for SIP NAP with g711 and g723 codecs]*______

3. Click on ***"default_g711_g723"***

a) Click on ***"SDP"***

i. In ***"Profile SDP Description"***, replace the first line by ***"m=audio 0 RTP/AVP 8 4 101 13"***
✍ For more information, refer to TelcoBridges Wiki, search for "Profile SDP Description"

✔ **Step 3 -** Configure "SBC-1" NAP with new profile

1. Click on ***"NAPs"*** menu
2. Click on ***"SBC_1"***

a) In ***"Default Profile"***, select ***"default_g711_g723"***

b) Click on ***"Save"***

☛ Validate your Results

- Click on ***"Profile"*** menu, then ***"default_g711_g723"*** and ***"SDP"*** to expand the field.
- Check the ***"Profile SDP Description"***:

```
m=audio 0 RTP/AVP 8 4 101 13
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-36
```

- Click on ***"NAPs"*** menu to verify:

Network Access Point List

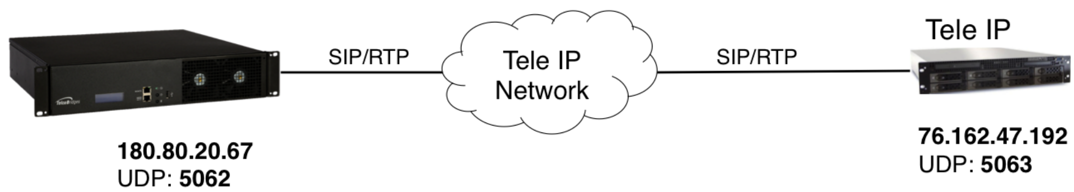
Name	Profile	Channel Usage	SIP Proxy	Members
SBC_1	default_g711_g723		UDP 192.168.101.77:5060	TS_101_160_5060, UNIT01:sip_101_160:pr_rtp_202_160

3.3 Recap

3.3.1 Exercise *(Optional)*

⌚ 30 minutes

Configure IP and SIP according to the following requirements:



1. "TeleIP" SIP peer
   - Local
     - SIP: 180.80.20.67, UDP port 5062
     - RTP: 180.80.20.67, UDP ports 20000-40000
     - Codecs: G.729ab and G.711 uLaw
     - Relay events 0 to 15 and 32 to 36 (RFC2833) with Payload Type 101
     - Voice activity detection (Confort Noise) with Payload Type 13
   - Remote
     - SIP: 76.162.47.192, port 5063, UDP
     - RTP: 76.162.47.192, UDP ports 19000-21000

✍ You must have completed the optional exercise of Section ***"IP configuration"*** in Lesson ***"IP Network"***

✎ Answer the Questions

1. How many SIP Transport Servers will you need? ______*[A8] [1 Transport server for IP 180.80.20.67 port 5062]*______
2. Do you need a new Profile? ______*[A9] [yes]*______
3. How many NAPs must be created? ______*[A10] [1 for Tele IP 76.162.47.192 port 5063]*______

➤ Start the Exercise

1. Select your configuration
2. Configure all the required components
3. Validate your configuration from Appendix subsection B.1.2 on page 9

APPENDIX **A**

References

A.1 QoS Reference

Quality Of Service (QoS) table showing the value translation between Type Of Services (ToS), Differentiated Service Code Point (DSCP) and Class of Service (CoS).

QoS Values Calculator v2

CoS = Class of Service
DSCP = Differentiated Services Code Point
ToS = Type of Service
AF = Assured Forwarding
IPP = IP Precedence
CS = Class Selector
DP = Drop Probability
ECN = Explicit Congestion Notification

	8th bit	7th bit	6th bit	5th bit	4th bit	3rd bit	2nd bit	1st bit
	ToS							
	DSCP						ECN	
	AF (CS;DP)							
	IPP=CS			DP				
				Delay	Thruput	Reliability		
ToS	128	64	32	16	8	4	2	1
DSCP	32	16	8	4	2	1		
CoS=IPP	4	2	1					

CoS=IPP	AF	DSCP	ToS	ToS HEX	DP	8th bit	7th bit	6th bit	5th bit	4th bit	3rd bit	2nd bit	1st bit
1	CS1	8	32	20		0	0	1	0	0	0	0	0
1	AF11	10	40	28	Low	0	0	1	0	1	0	0	0
1	AF12	12	48	30	Medium	0	0	1	1	0	0	0	0
1	AF13	14	56	38	High	0	0	1	1	1	0	0	0
2	CS2	16	64	40		0	1	0	0	0	0	0	0
2	AF21	18	72	48	Low	0	1	0	0	1	0	0	0
2	AF22	20	80	50	Medium	0	1	0	1	0	0	0	0
2	AF23	22	88	58	High	0	1	0	1	1	0	0	0
3	CS3	24	96	60		0	1	1	0	0	0	0	0
3	AF31	26	104	68	Low	0	1	1	0	1	0	0	0
3	AF32	28	112	70	Medium	0	1	1	1	0	0	0	0
3	AF33	30	120	78	High	0	1	1	1	1	0	0	0
4	CS4	32	128	80		1	0	0	0	0	0	0	0
4	AF41	34	136	88	Low	1	0	0	0	1	0	0	0
4	AF42	36	144	90	Medium	1	0	0	1	0	0	0	0
4	AF43	38	152	98	High	1	0	0	1	1	0	0	0
5	CS5	40	160	A0		1	0	1	0	0	0	0	0
5	EF	46	184	B8		1	0	1	1	1	0	0	0
6	CS6	48	192	C0	Routing	1	1	0	0	0	0	0	0
7	CS7	56	224	E0	Network	1	1	1	0	0	0	0	0

Version:
v2 - ToS in HEX added

NetContractor

Reference link: `http://www.netcontractor.pl/blog/?p=371`

APPENDIX B

Exercise Expected Results

B.1 SIP

B.1.1 SIP Configuration

Click on ***"SIP"*** menu, and then click on ***"sip_UNIT01"*** to verify:

Transport Servers

Name	Port	Port Type	IP interfaces
TS_20_67_5070	5070	UDP	public_20_67

Click on ***"NAPs"*** menu to verify:

Network Access Point List

Name	Profile	Channel Usage	SIP Proxy	Members
IP_World	default		UDP 80.65.32.64:5060	TS_20_67_5070, UNIT01:public_20_67:pr_public_20_67

B.1.2 SIP Recap

Click on ***"SIP"*** menu, and then click on ***"sip_UNIT01"*** to verify:

Transport Servers

Name	Port	Port Type	IP interfaces
TS_20_67_5062	5062	UDP	public_20_67

Click on ***"NAPs"*** menu to verify:

Network Access Point List

Name	Profile	Channel Usage	SIP Proxy	Members
TeleIP	default_g729_g711		UDP 76.162.47.192:5063	TS_20_67_5062, UNIT01:public_20_67:pr_public_20_67_20k_30k

Click on ***"Profile"*** menu, verify the G.729/G.711 ***"Profile SDP Description"***:

```
m=audio 0 RTP/AVP 18 0 101 13
a=rtpmap:101 telephone-event/8000
a=fmtp:101 0-15,32-36
```