How to setup SIP Trunking ?

SEUK

SIP Trunk Providers

- This is a list of the current SIP Trunk providers that have been tested on the OfficeServ 7000 range:
  - Gamma – IP Direct V3.0 and V3.1 versions
  - VoiceFlex
  - VoIP Unlimited
  - Coms.com
  - CLUB-COMMS
  - AQL
  - Thus
  - O-bit
  - iHub
- Please note that only one SIP service provider can be used on the systems at any one time.
- Although the above suppliers have been tested you should be made aware that providers periodically change their service.

Software Levels

- For SIP Trunking only the below are supported:
  - OfficeServ 7400 – V3.34 and above
  - OfficeServ 7200 – V2.69 and above
  - OfficeServ 7100 – V4.07 and above
  - OfficeServ 7030 - All versions
  - OfficeServ 500 - V2.69
  - OfficeServ 100 - V2.69
- Please note that there is no further development for the OfficeServ 100 and 500

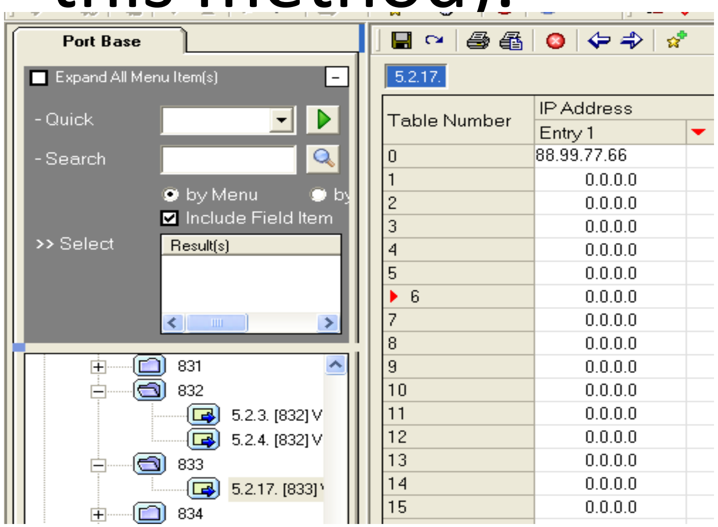
Please note !

- The following screen shots where taken using an OS 7200 running v4.30a and v1.30a of the Installation Tool. Both are available to download from the Support website.

IP address authentication method MMC’s 832 & 833

MMC 833

- Some SIP Trunk providers use a simple setup using the Public IP as the authentication method
- MMC 833 is used to route SIP calls to a specified IP address (O-bit,AQL,CLUB-COMMS and Thus use this method).



MMC 833 enter SIP providers IP Address

MMC 832

- All numbers route to IP table 0 (IP address of trunk provider)

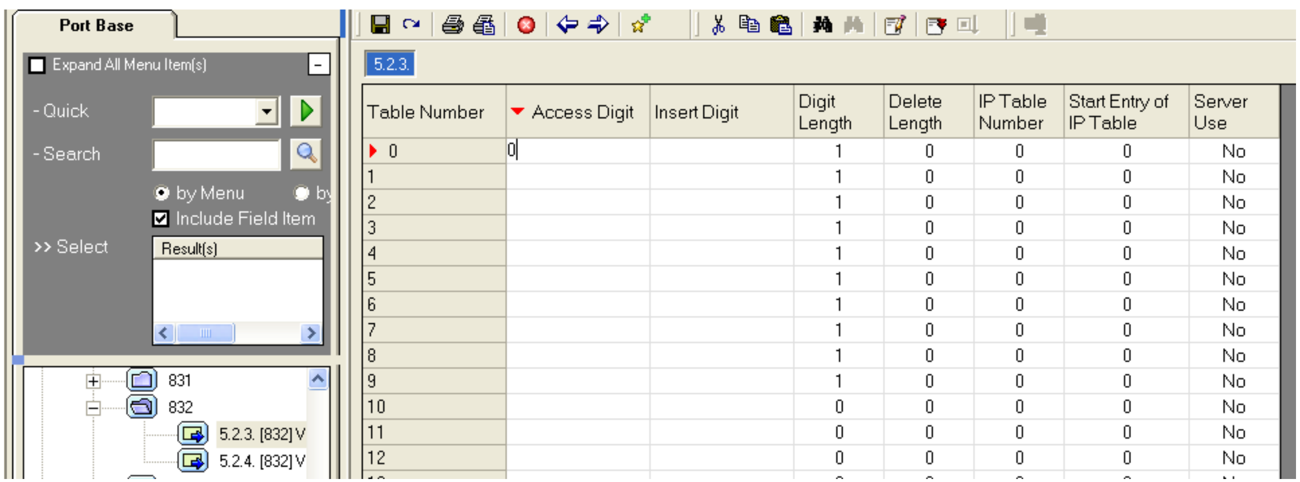


Table Number	Access Digit	Insert Digit	Digit Length	Delete Length	IP Table Number	Start Entry of IP Table	Server Use
0	0		1	0	0	0	No
1			1	0	0	0	No
2			1	0	0	0	No
3			1	0	0	0	No
4			1	0	0	0	No
5			1	0	0	0	No
6			1	0	0	0	No
7			1	0	0	0	No
8			1	0	0	0	No
9			1	0	0	0	No
10			0	0	0	0	No
11			0	0	0	0	No
12			0	0	0	0	No

MMC 837 – SIP Options

- Other providers require information to be sent to them in the form of a username and password
- In each of the following you will see that a username and password is required. These should be supplied to you by the SIP trunk provider along with the IP address and domain name
- If you are not sure or you provider is not covered here please contact support

MMC 837 – Gamma example

Item	Value
SIP Carrier Name	Gamma v3.0
SIP Server Enable	Enable
SIP Service Available	No
Registra Address	88.215.60.40
Registra Port	5060
Outbound Proxy	88.215.60.40
Alternative Outband Proxy	0.0.0.0
Outbound Proxy Port	5060
Proxy Domain Name	gw3.theiptele.com
User Name	username
Auth Username	username
Auth Password	password
Regist. Per User	Disable
Session Timer	None
Session Expire Time	1800
Trunk Reg Expire Time	30
Alive Notify	None
Alive Notify Time	1800
Carrier Exclusive	Disable
IMS Option	Disable
P-Asserted-ID Use	None
Privacy	Disable
SIP Peering	Disable
Send CLI Table	1
Supplementary Type	Pbx managed
302 Response	Disable
SIP User Table ID	1
SIP Destination Type	To Header
Codec Auto Nego	Enable

Using this method Gamma will issue you with a username and password

MMC 837 – VoIP Unlimited

Item	Value
SIP Carrier Name	VoIP Unlimited
SIP Server Enable	Enable
SIP Service Available	No
Registra Address	91.151.2.130
Registra Port	5060
Outbound Proxy	91.151.2.130
Alternative Outband Proxy	0.0.0.0
Outbound Proxy Port	5060
Proxy Domain Name	sip.voip-unlimited.net
User Name	username
Auth Username	username
Auth Password	password
Regist. Per User	Disable
Session Timer	None
Session Expire Time	1800
Trunk Reg Expire Time	30
Alive Notify	None
Alive Notify Time	1800
Carrier Exclusive	Disable
IMS Option	Disable
P-Asserted-ID Use	None
Privacy	Disable
SIP Peering	Disable
Send CLI Table	1
Supplementary Type	Pbx managed
302 Response	Disable
SIP User Table ID	1
SIP Destination Type	To Header
Codec Auto Nego	Enable

MMC 837 – VoiceFlex

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Item	Value
SIP Carrier Name	VoiceFlex
SIP Server Enable	Enable
SIP Service Available	No
Registra Address	146.101.248.200
Registra Port	5060
Outbound Proxy	146.101.248.200
Alternative Outband Proxy	0.0.0.0
Outbound Proxy Port	5060
Proxy Domain Name	sip.voiceflex.com
User Name	username
Auth Username	username
Auth Password	password
Regist. Per User	Disable
Session Timer	None
Session Expire Time	1800
Trunk Reg Expire Time	30
Alive Notify	None
Alive Notify Time	1800
Carrier Exclusive	Disable
IMS Option	Disable
P-Asserted-ID Use	None
Privacy	Disable
SIP Peering	Disable
Send CLI Table	1
Supplementary Type	Pbx managed
302 Response	Disable
SIP User Table ID	1
SIP Destination Type	To Header
Codec Auto Nego	Enable

MMC 837 – Coms.com

Item	Value
SIP Carrier Name	COMS.COM
SIP Server Enable	Enable
SIP Service Available	Yes
Registra Address	85.90.225.100
Registra Port	5060
Outbound Proxy	85.90.225.100
Alternative Outband Proxy	0.0.0.0
Outbound Proxy Port	5060
Proxy Domain Name	sip.coms.com
User Name	username
Auth Username	username
Auth Password	password
Regist. Per User	Disable
Session Timer	None
Session Expire Time	20
Trunk Reg Expire Time	1800
Alive Notify	None
Alive Notify Time	1800
Carrier Exclusive	Disable
IMS Option	Disable
P-Asserted-ID Use	None
Privacy	Disable
SIP Peering	Disable
Send CLI Table	1
Supplementary Type	Samsung
302 Response	Disable
SIP User Table ID	1
SIP Destination Type	To Header
Codec Auto Nego	Enable

MMC 837 – iHub

Item	Value
SIP Carrier Name	iHub
SIP Server Enable	Enable
SIP Service Available	Yes
Registra Address	0.0.0.0
Registra Port	5060
Outbound Proxy	194.0.147.70
Alternative Outband Proxy	0.0.0.0
Outbound Proxy Port	5060
Proxy Domain Name	sippbx.hostedipt.co.uk
User Name	slgtstpan
Auth Username	slgtstpan
Auth Password	slg1234
Regist. Per User	Disable
Session Timer	None
Session Expire Time	1800
Trunk Reg Expire Time	3600
Alive Notify	None
Alive Notify Time	1800
Carrier Exclusive	Disable
IMS Option	Disable
P-Asserted-ID Use	None
Privacy	Disable
SIP Peering	Disable
Send CLI Table	1
Supplementary Type	Samsung
302 Response	Disable
SIP User Table ID	1
SIP Destination Type	Request-URI
Codec Auto Nego	Enable

Trunk Numbers & Groups

- All SIP calls will require a MGI channels if you are using Keyphones or POT's on your system
  - MGI Trunks are numbered 3801 onwards
  - SIP Trunks have a virtual number of 8501 onwards
  - This number scheme can be adjusted in MMC 724 as normal
- By default all SIP trunks have been put into Trunk Group 805 (Group 802 on the OS7030)
  - i.e. Dial 805 to dial out over SIP

Routing outgoing calls

- Calls can be sent out of the system using the LCR tables

3.1.2.

Entry Number	LCR Digit	Length	Route Table
1	01	5	1
2	02	5	1
3	07	5	1
4			

3.1.4.

Class No	Zone 1		Zone 2		Zone
	Group	Modify	Group	Modify	Grou
1	805	1			
2					
3					

Use the SIP trunk group

3.1.5.

Entry Number	Delete Count	Insert Digits	Append Digits
1	0		#
2	0		
3	0		

Routing incoming calls

- All calls in to an OfficeServ 7000 will appear as DDI numbers
- These can be routed accordingly by programming MMC 714 – DDI Translation Table
- All incoming calls/DDI numbers not programmed in MMC 714 will be routed to the Operator/Group 500 by default

Out going CLI

- CLI is programmed as normal using MMC 323 – Send CLIP Number.
- However, this feature is supplier dependant
  - You should check your SIP service provider to confirm what CLIs (and format) can be sent

Network Setup

- Network setup should be treated the same as for any VoIP calls
- MMC 830/831 should be programmed as normal
  - Public and private IP address
  - System IP Type = Public with private
- Network router should be running NAT (if required)
  - Port 5060 udp should be forwarded to the MCP/MP processor
- Firewalls should allow port 5060 udp

Setting up a Trace

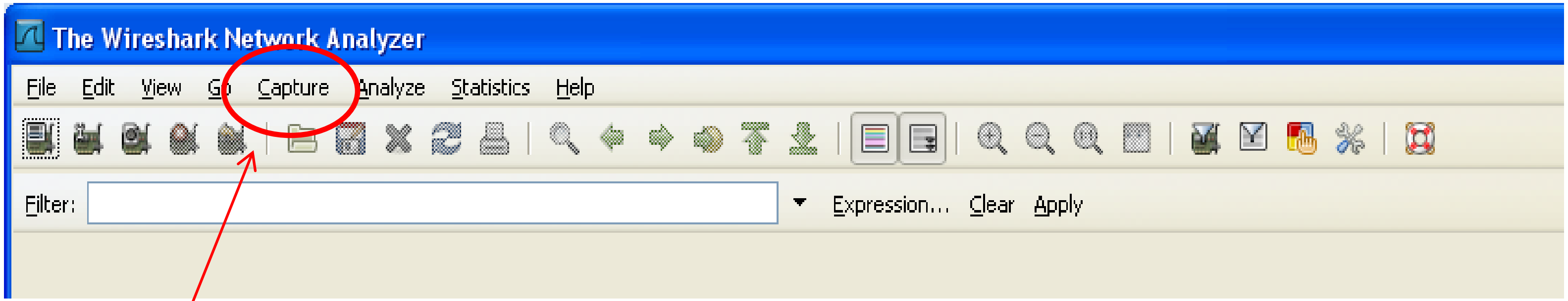
If there is a problem

- If you have a problem making or receiving calls we may ask you to provide us with a trace. This will involve capturing the SIP packets travelling to and from the system.
- This can be achieved using a network Hub (not a switch) and ‘packet sniffer’ software
- Wireshark is generally used for this purpose . It can be downloaded free of charge from http://www.wireshark.org

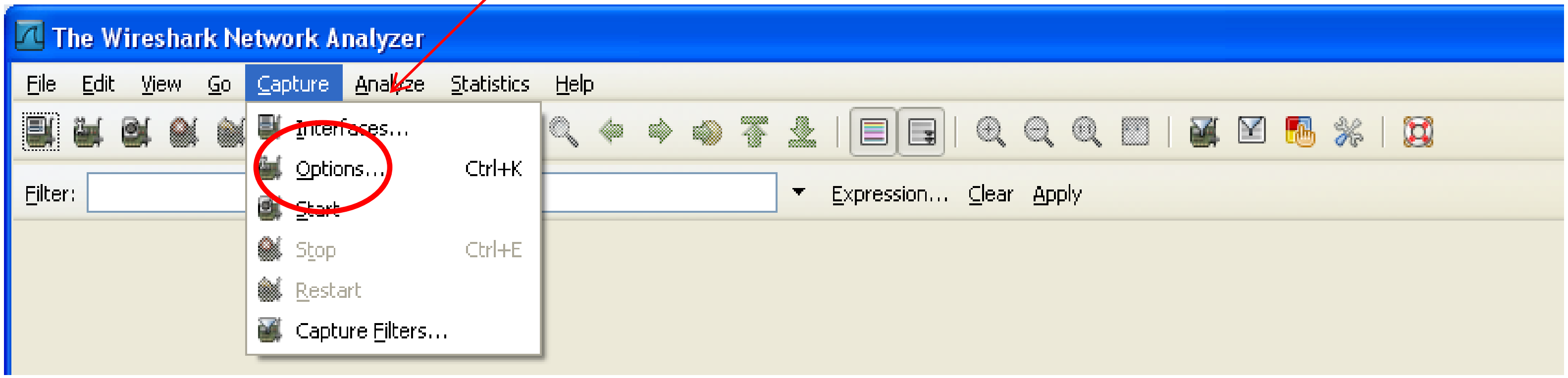
- Download and install Wireshark
- Set a trace running
- Attempt to make a call
- Capture the traffic in and out of the system

Connect the system the router and your laptop into the Hub

Setting up a trace in WireShark

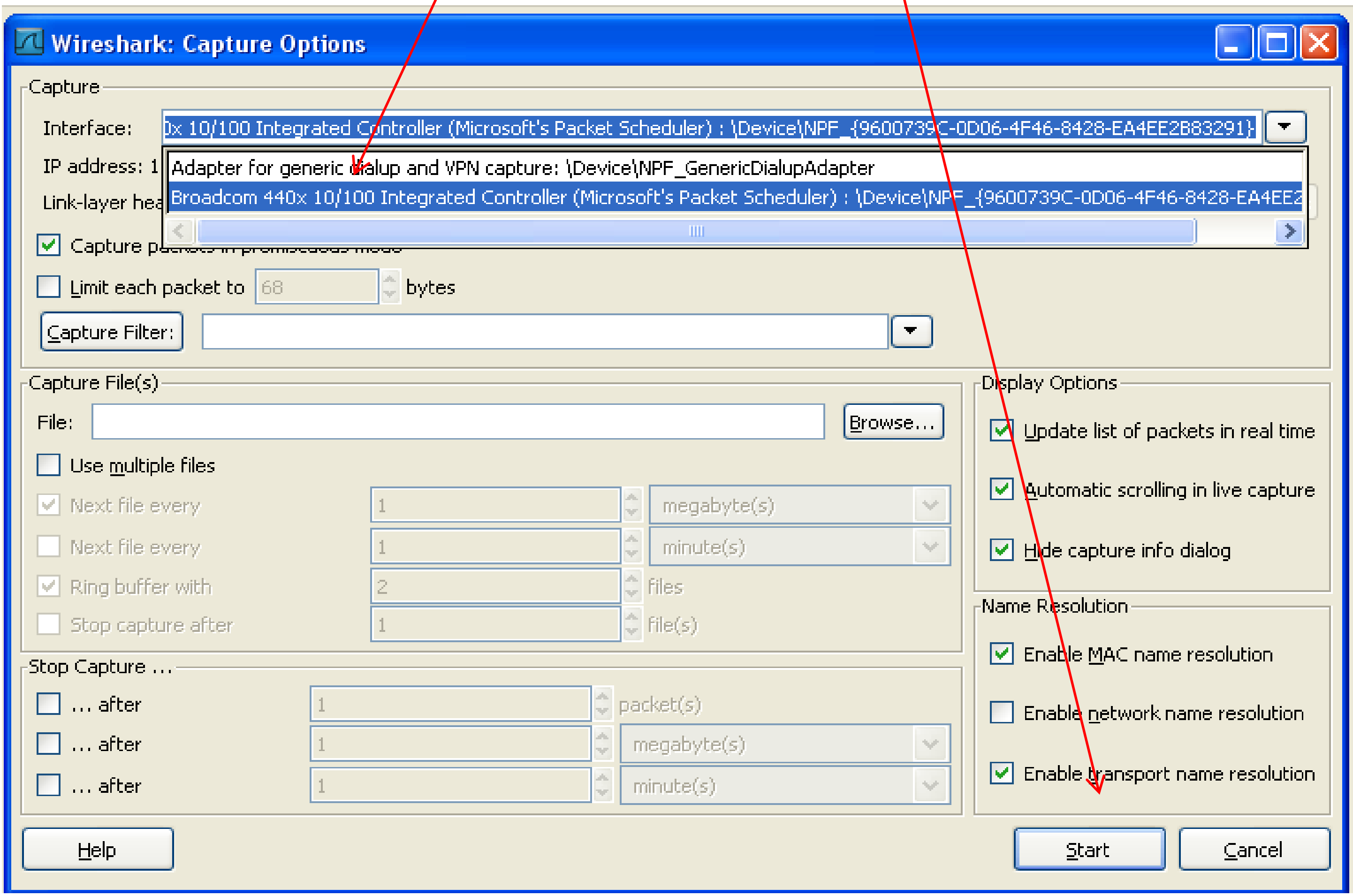


Click capture **click options**



Setting up a trace in WireShark

Select your PC's network interface then click start



Setting up a trace in WireShark

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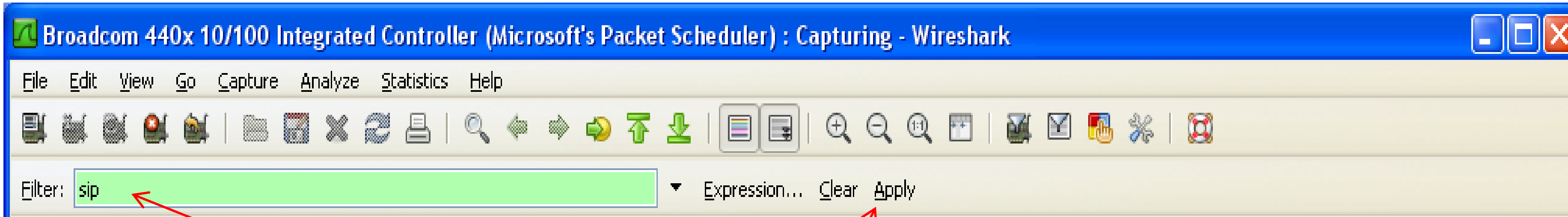
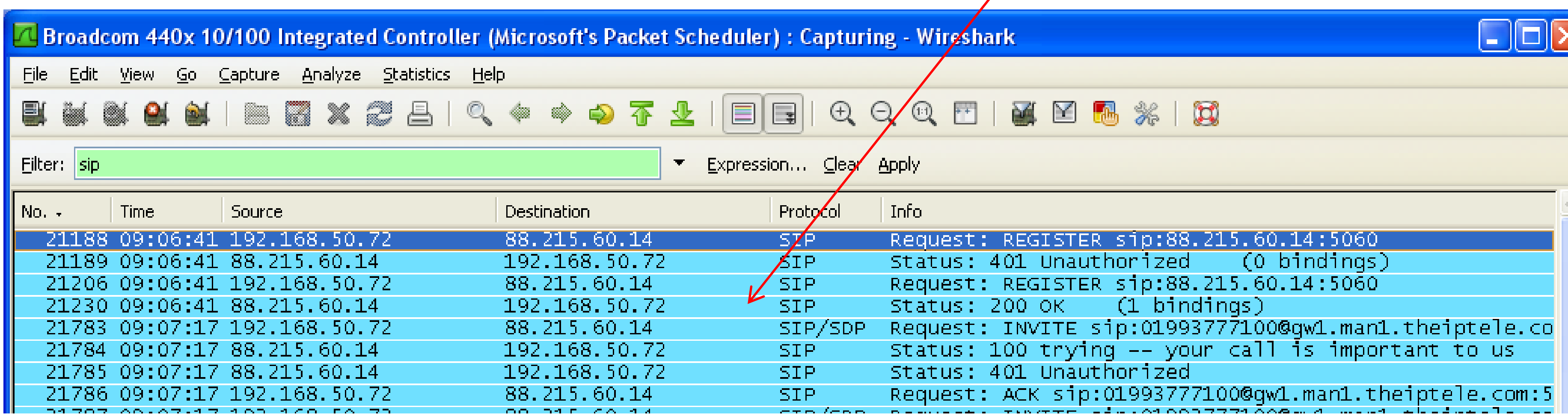
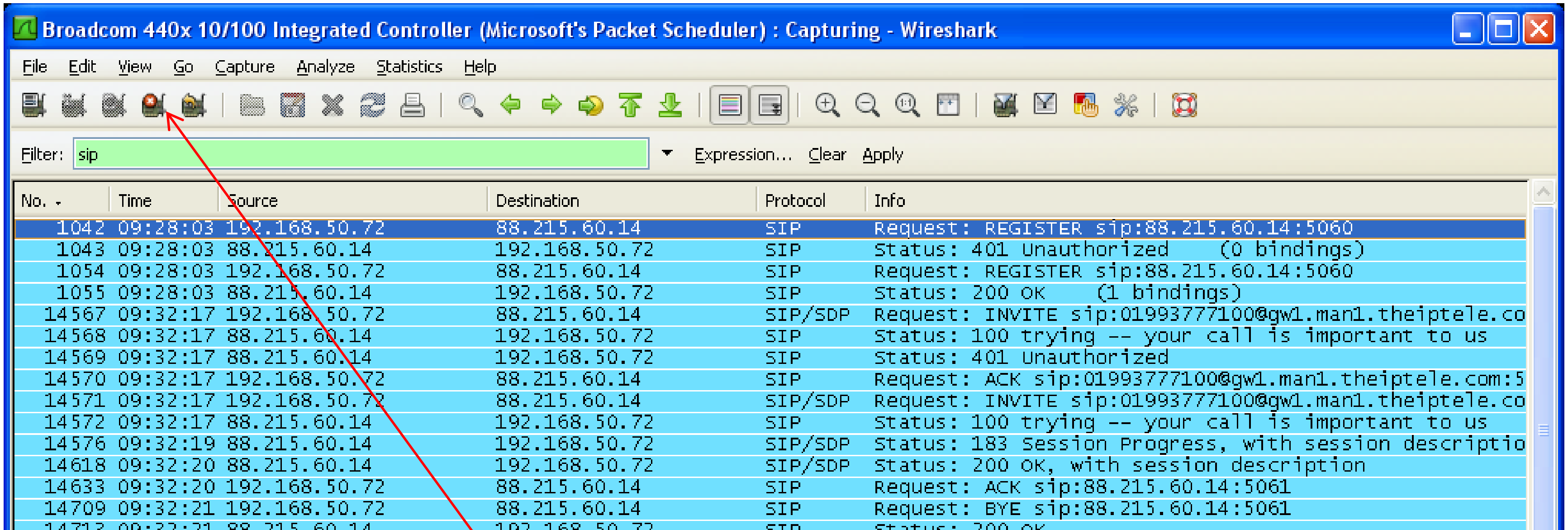


Type SIP into the filter box then click apply

You should then see SIP packets to and from as below

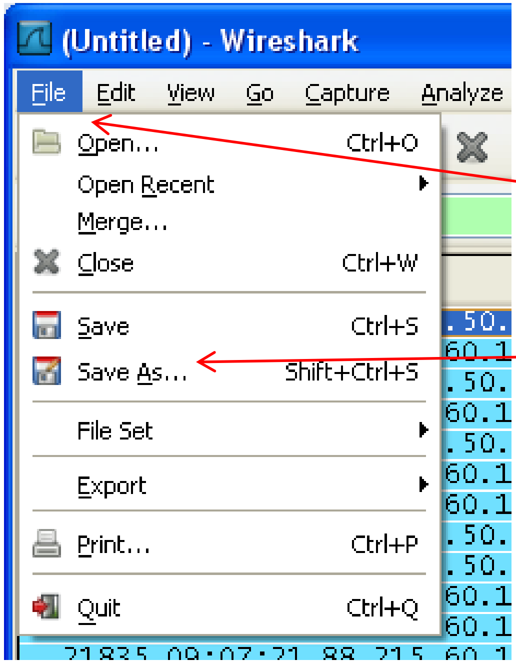


Setting up a trace in WireShark



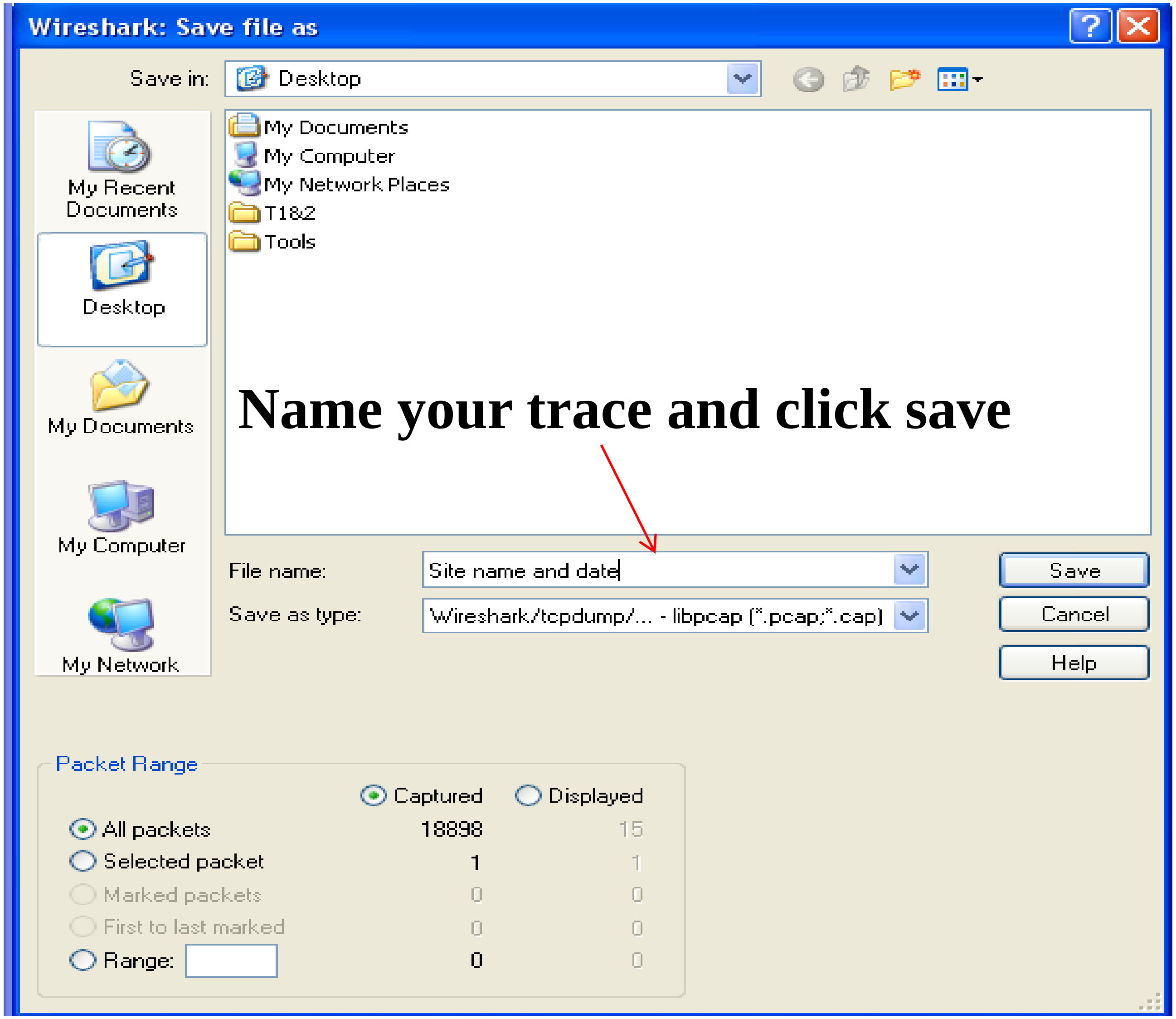
When you have captured an attempt to make a call click stop

Saving the trace



Click file then save as..

Saving the trace



- The file can now be attached to a ticket on the support website