VoIP Troubleshooting and Analysis

June 16th 2011

Loris Degioanni

Sr. Director of Technology | Riverbed Technology

Pietro Giordano

Technical Staff Team Lead | Riverbed Technology

SHARKFEST **'11**

Stanford University

June 13-16, 2011

VoIP Signaling and Data

- Signaling protocols
  - Manage the call
    - Notify a receiver for incoming calls
    - Notify the caller for accepted or rejected calls
    - Notify both the caller and the receiver for errors/events
  - Manage the data protocols
    - Decides how the voice is encoded
    - Decides how the voice is transferred
- Data protocols
  - Carry the coded voice

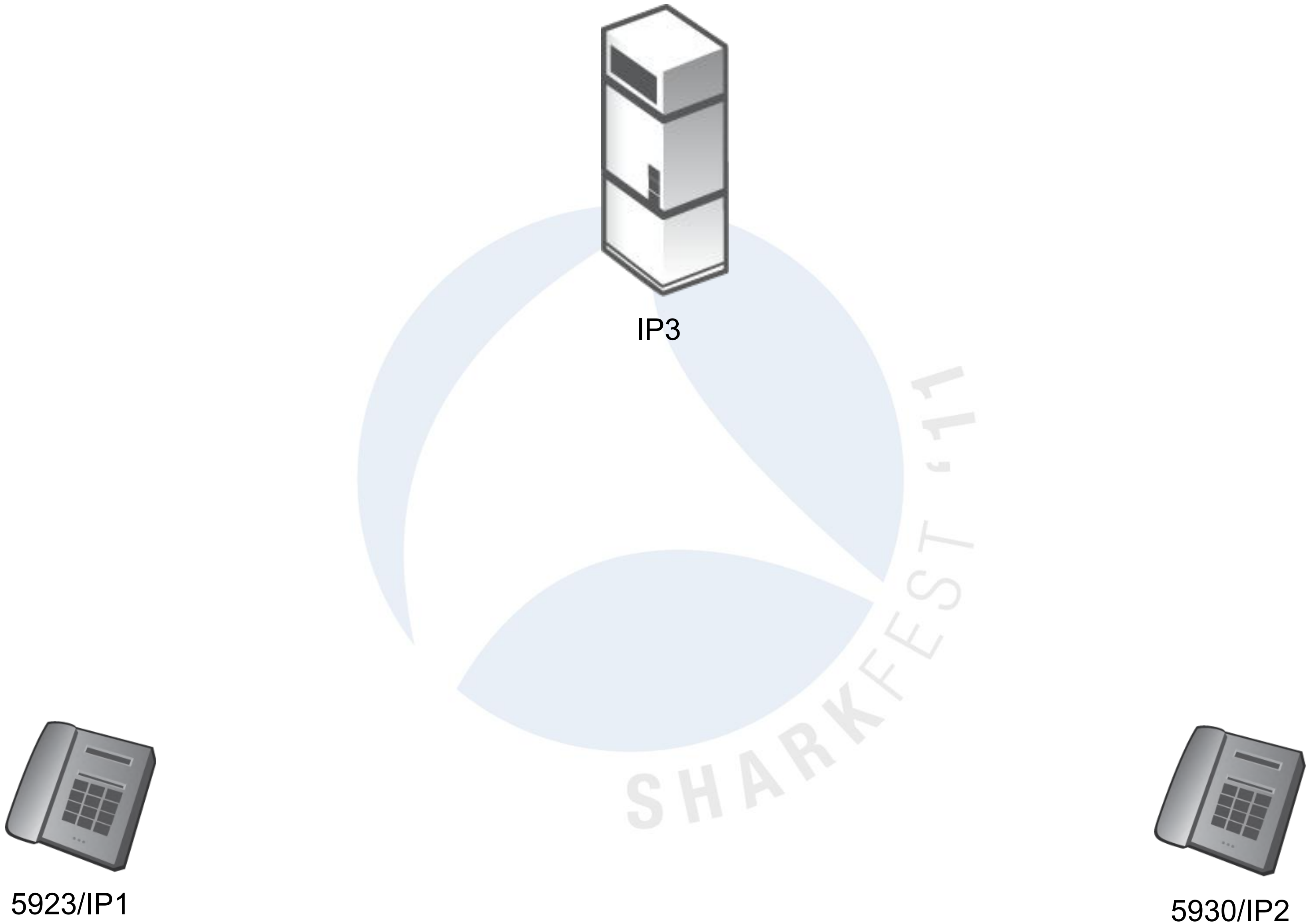
VoIP Signaling and Data

- Signaling protocols
  - SIP (IETF)
  - H323 (ITU-T)
  - SCCP or Skinny (CISCO)
  - MCGP (IETF)
  - Others
- Data protocols
  - RTP (PCMU, PCMA, G728, G729…)

VoIP Signaling and Data

- Cascade Pilot views (sip_ex_01_sig_data.pcap)
  - VoIP Signaling vs. Data Bandwidth Over Time
  - VoIP Signaling vs. Data Bandwidth Utilization
  - RTP Traffic by CODEC
  - Top RTP CODECs

VoIP call: a simple example



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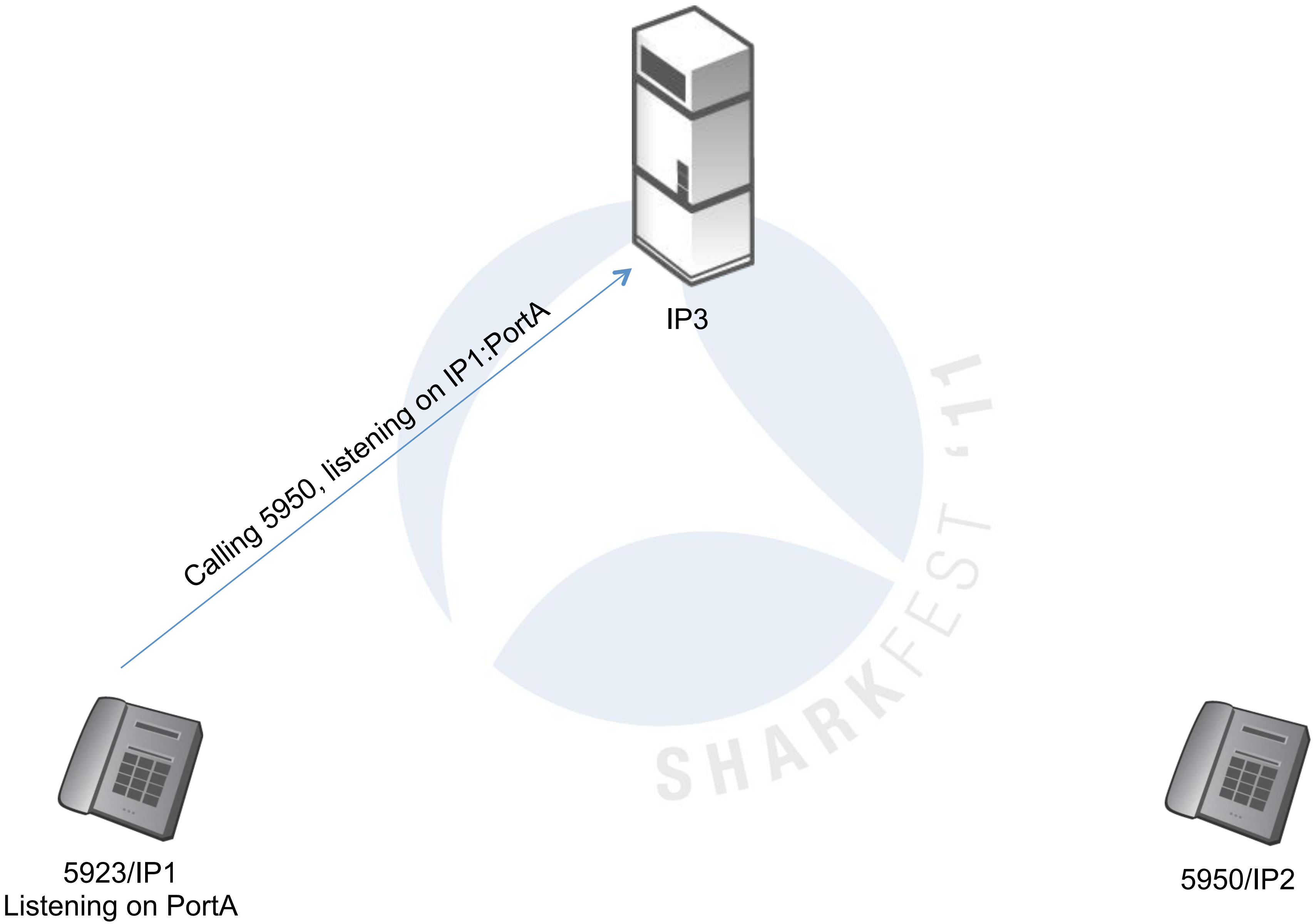

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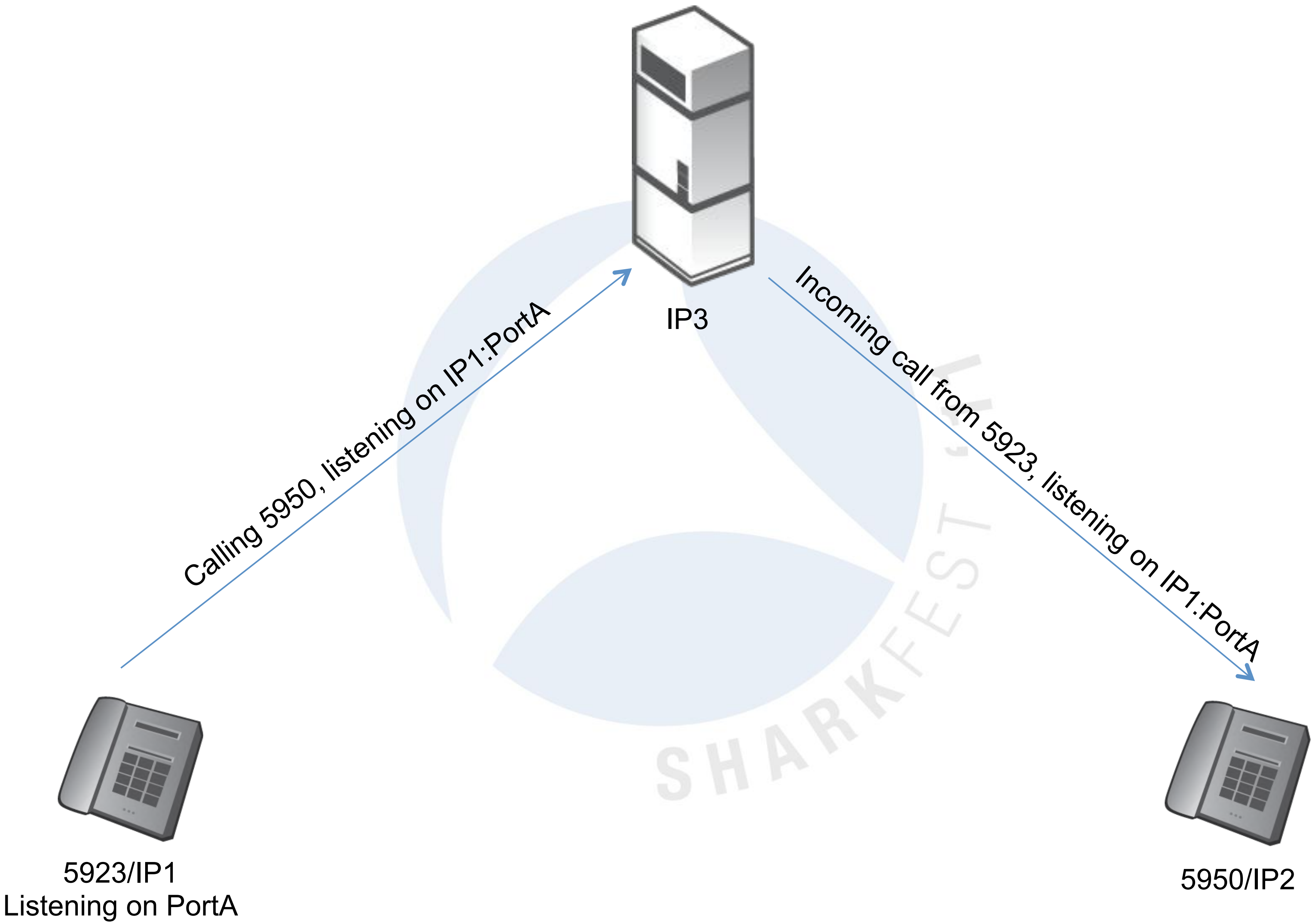

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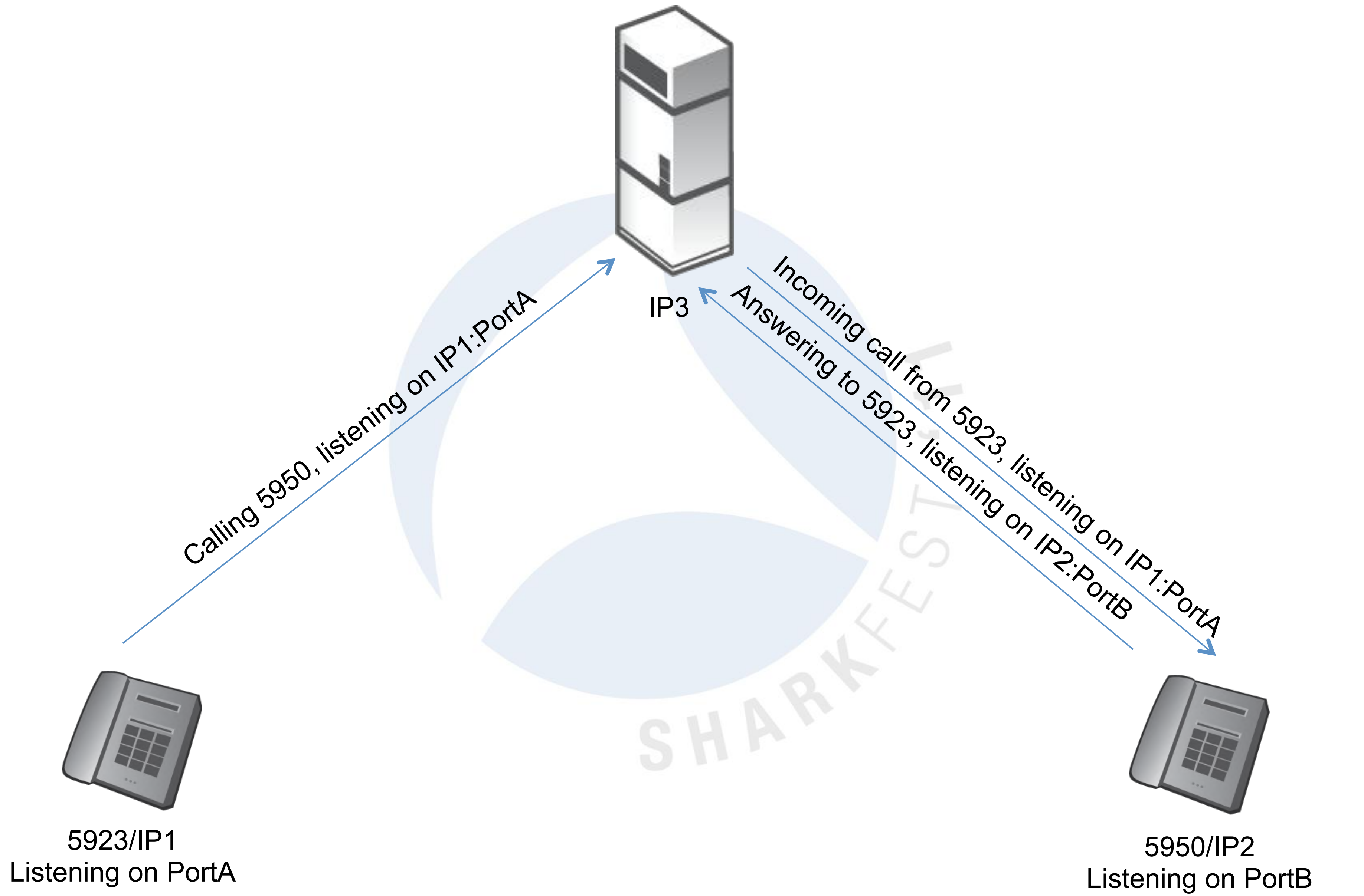

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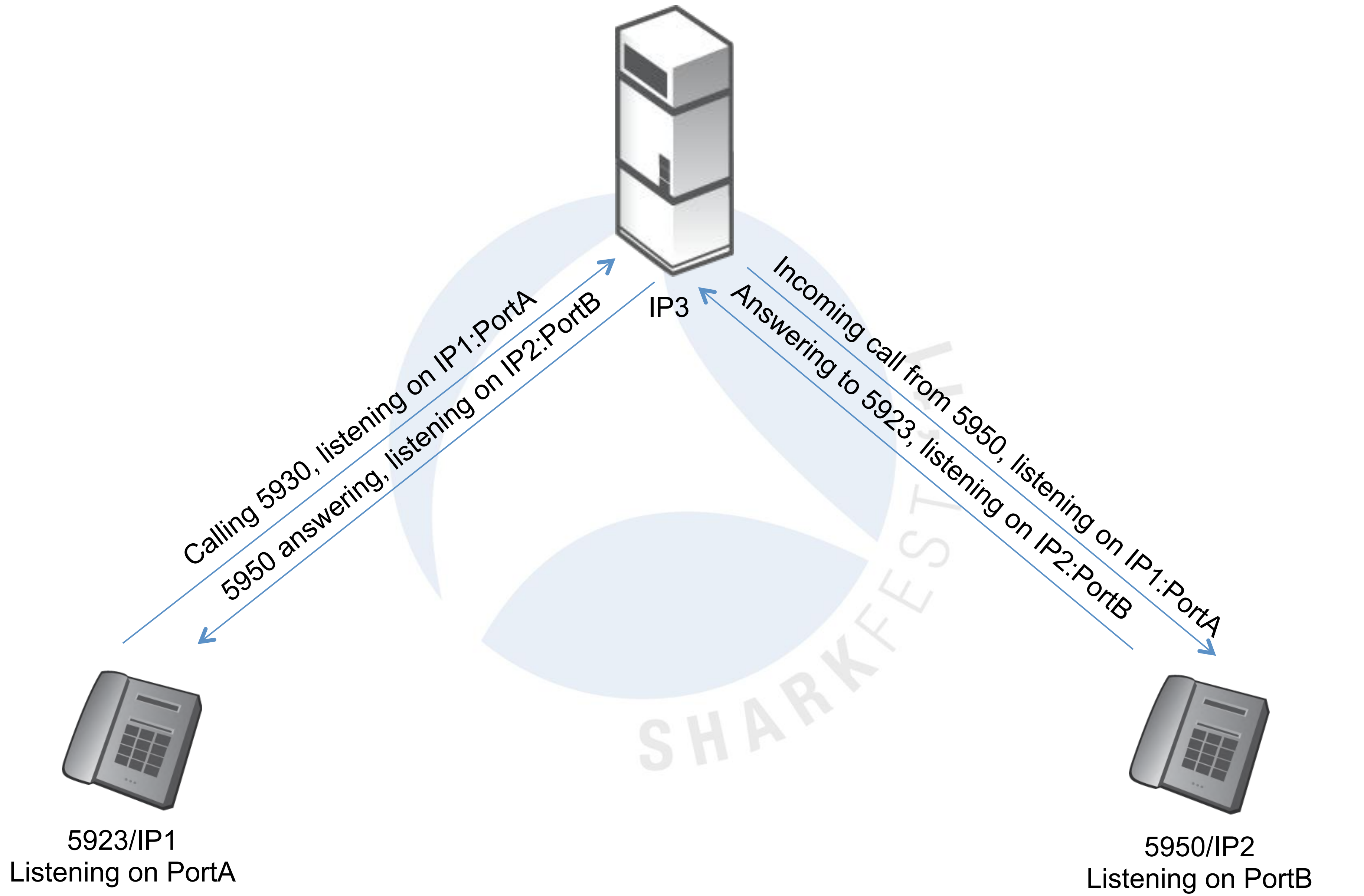

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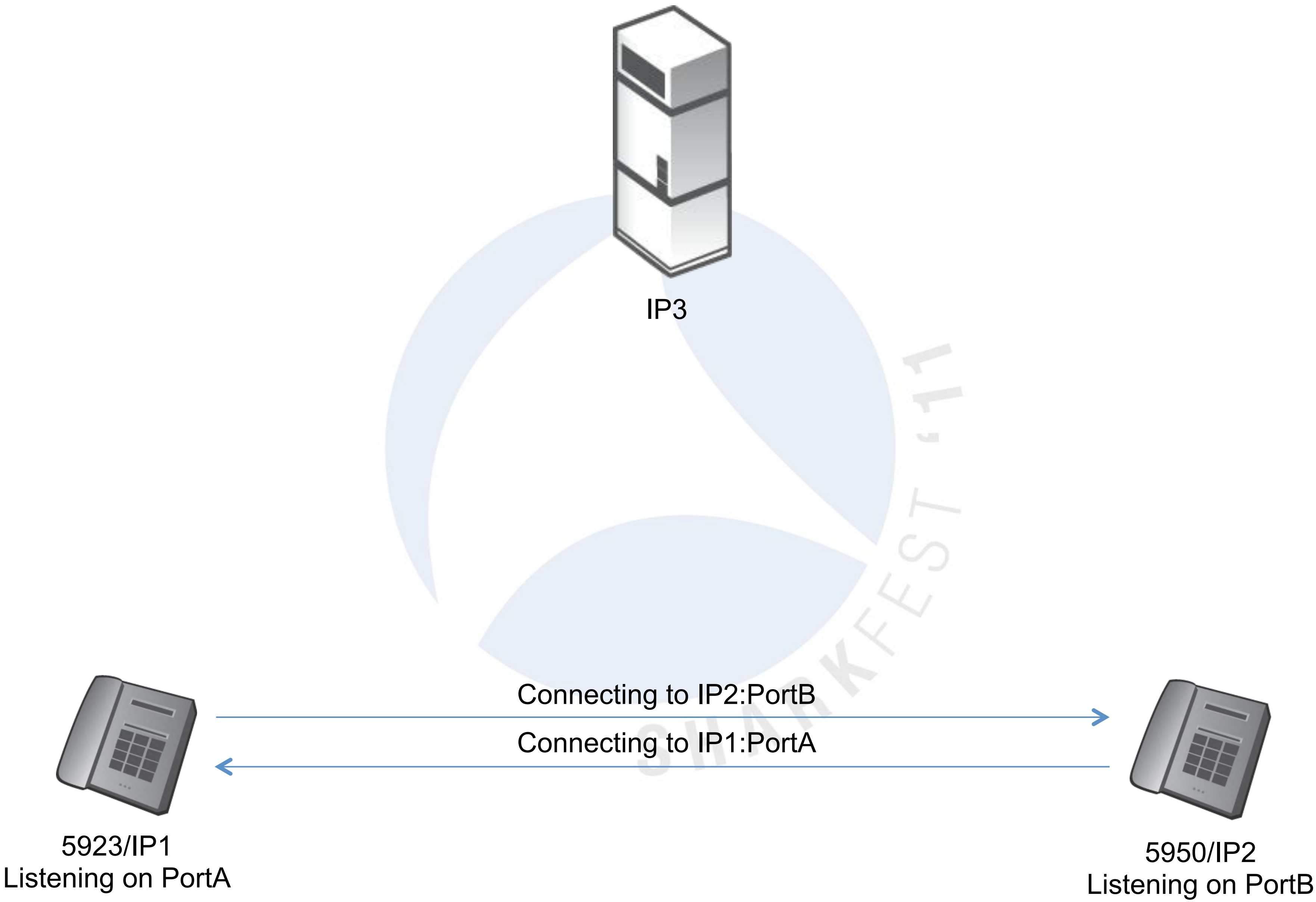


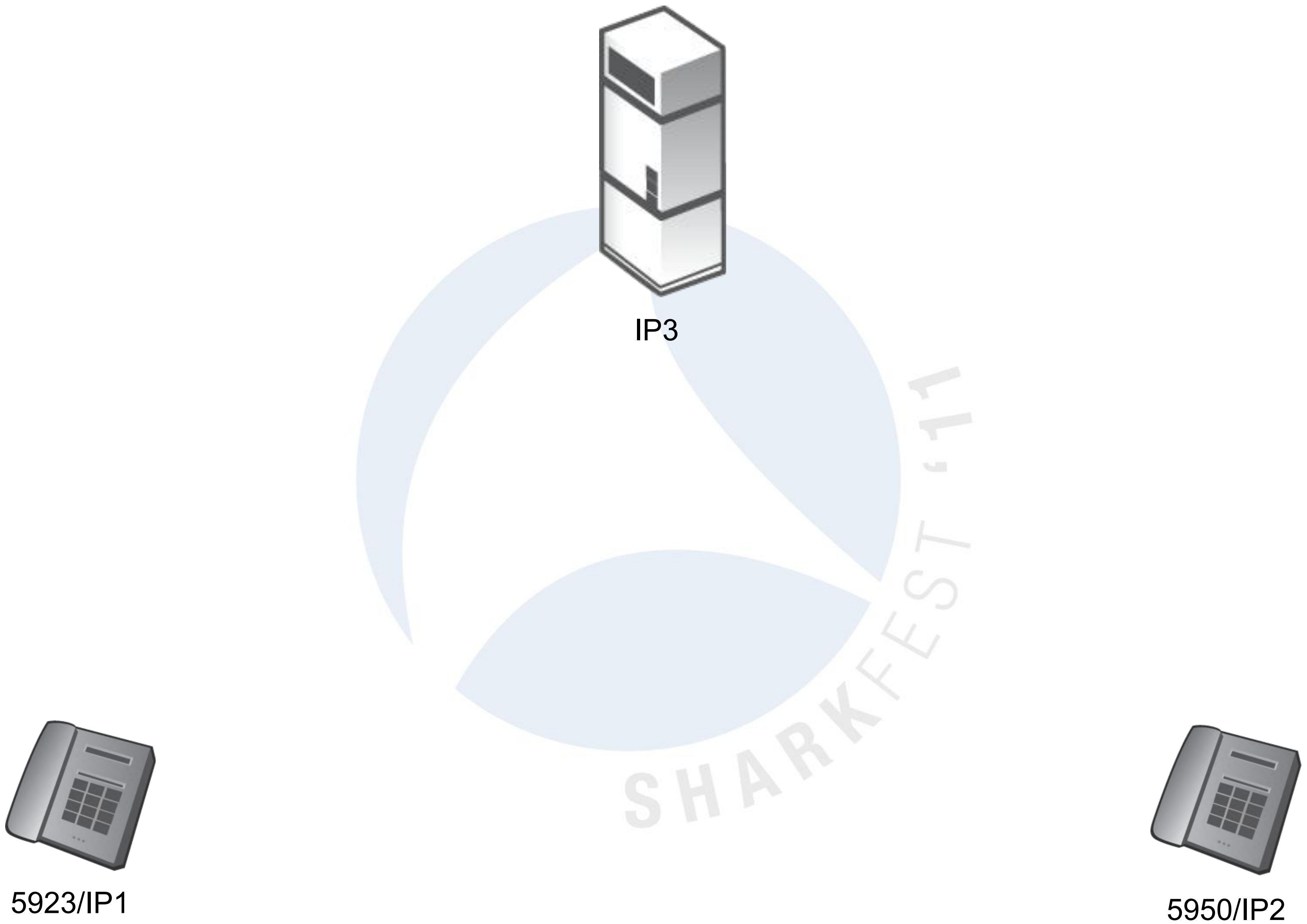
VoIP call: a simple example

– Cascade Pilot views (sip_ex_01_reg.pcap)
  - Transaction Analysis by VoIP Call

5923/IP1

5930/IP2

VoIP server as relay



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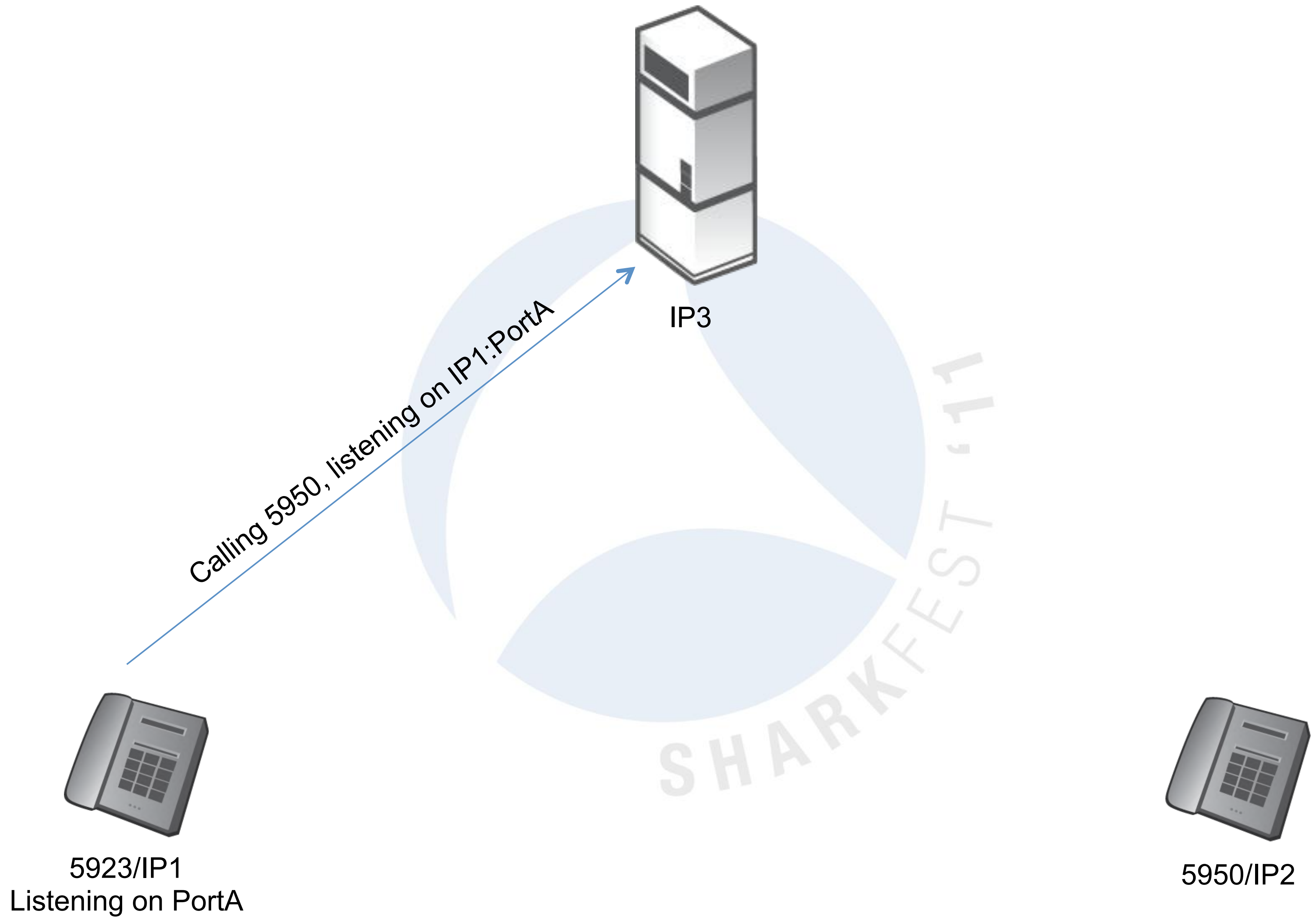

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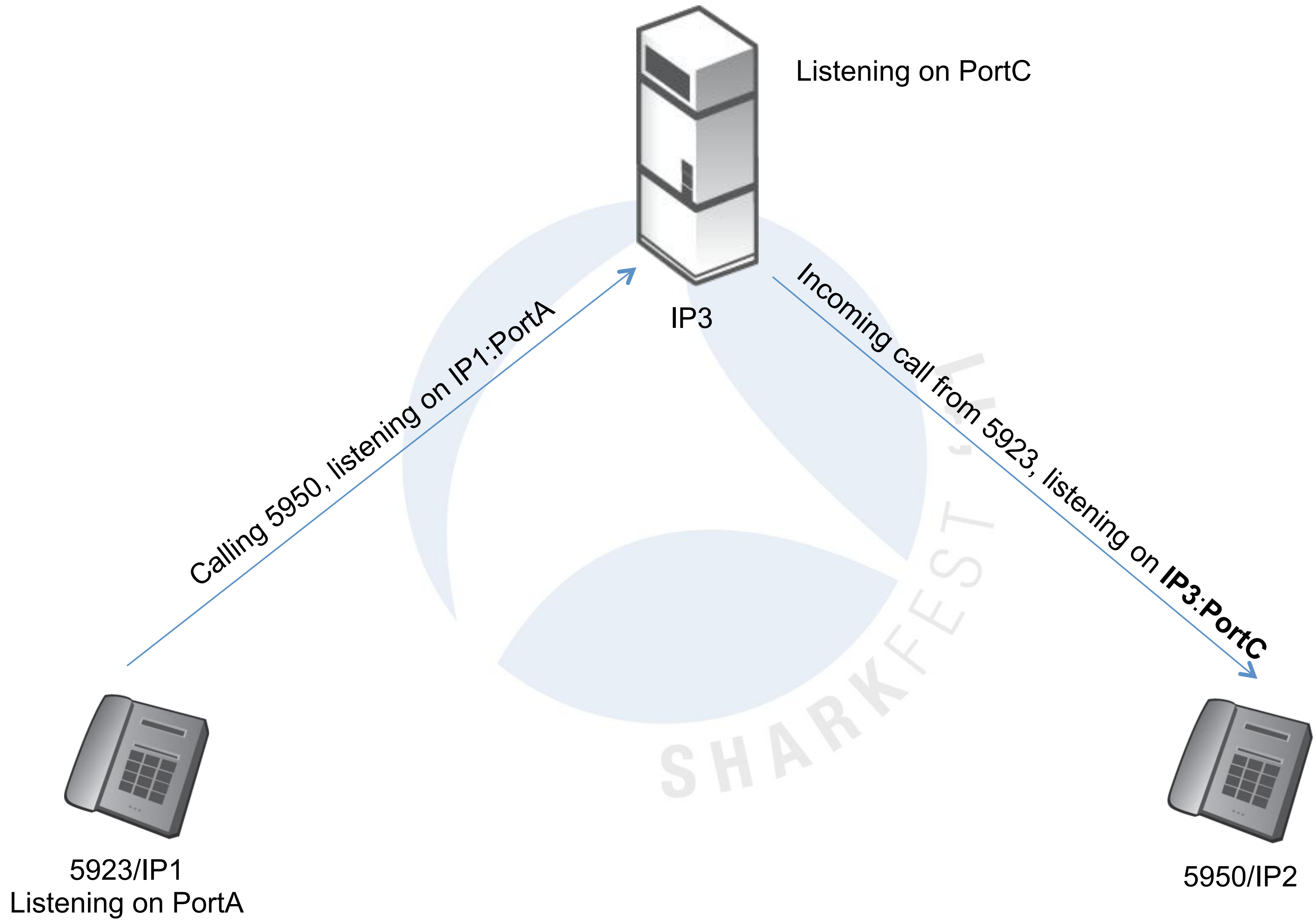

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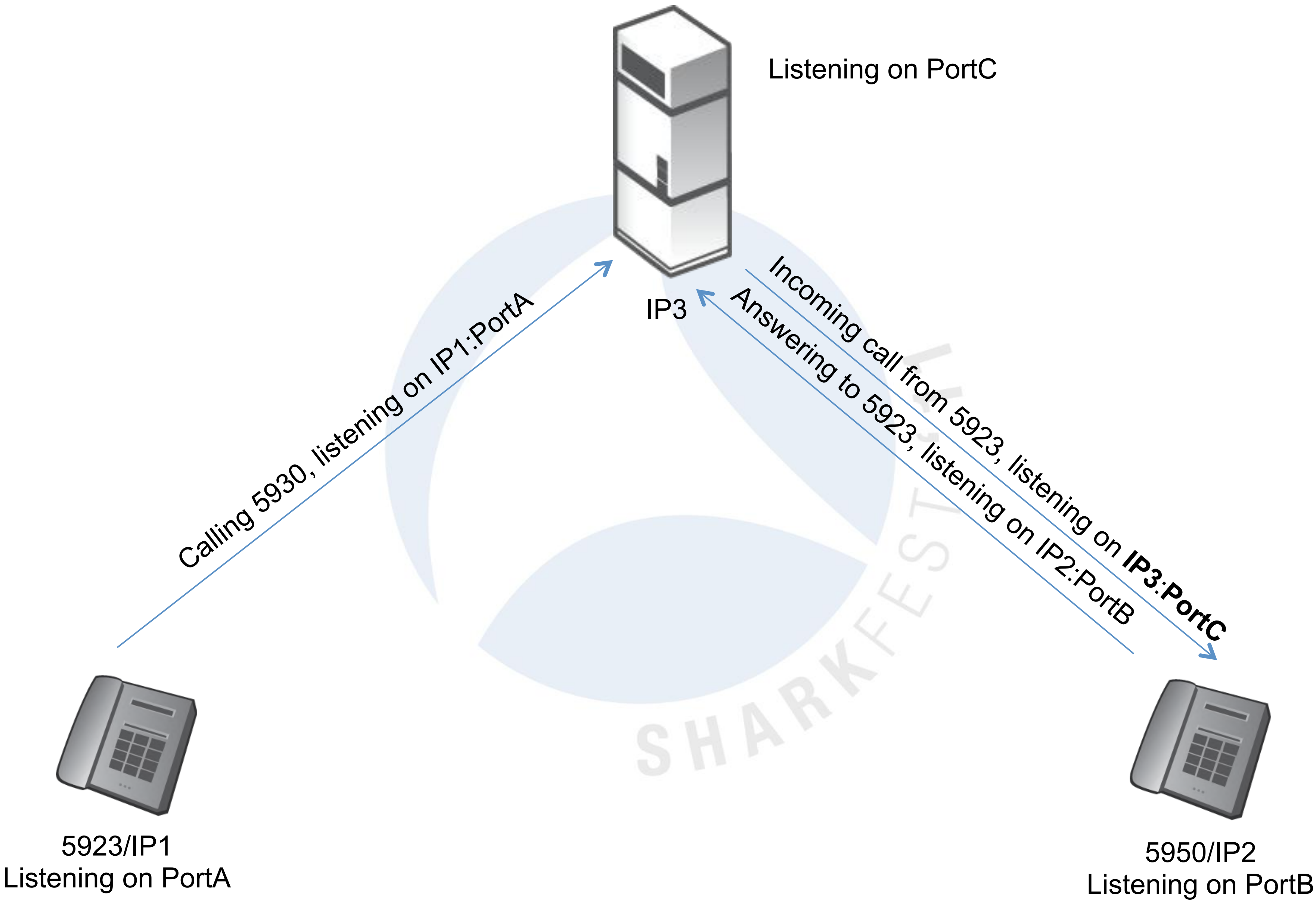

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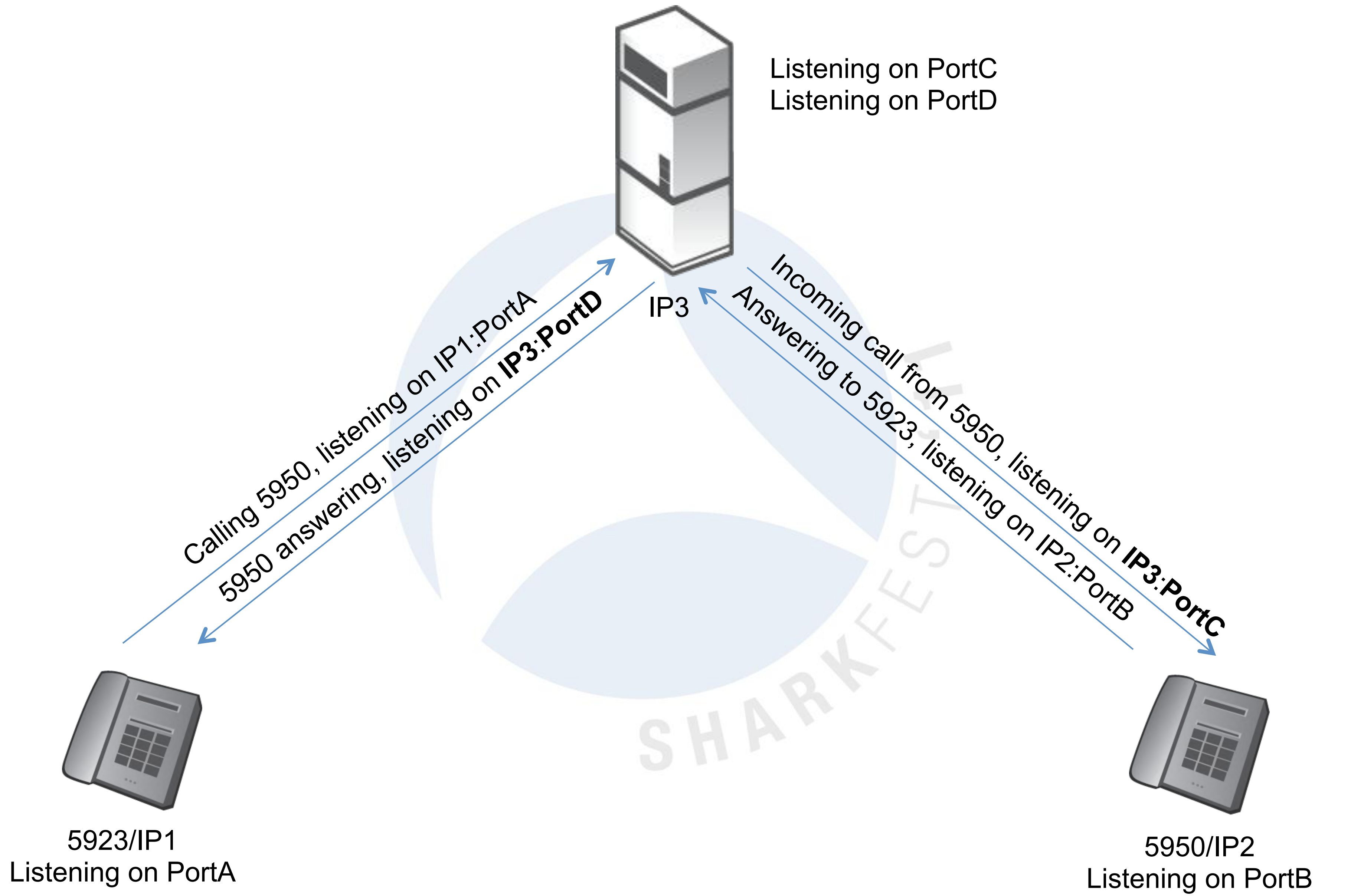

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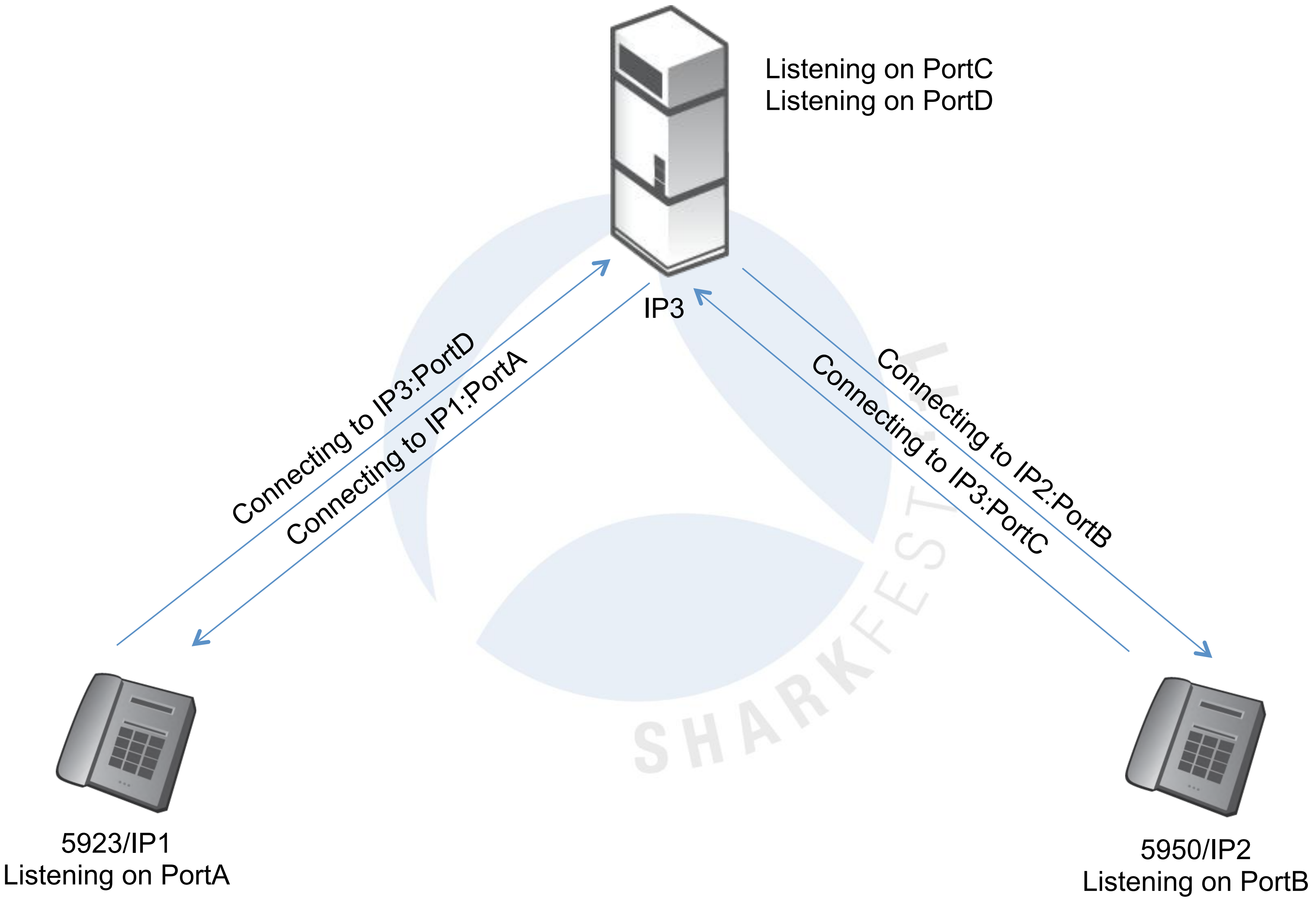
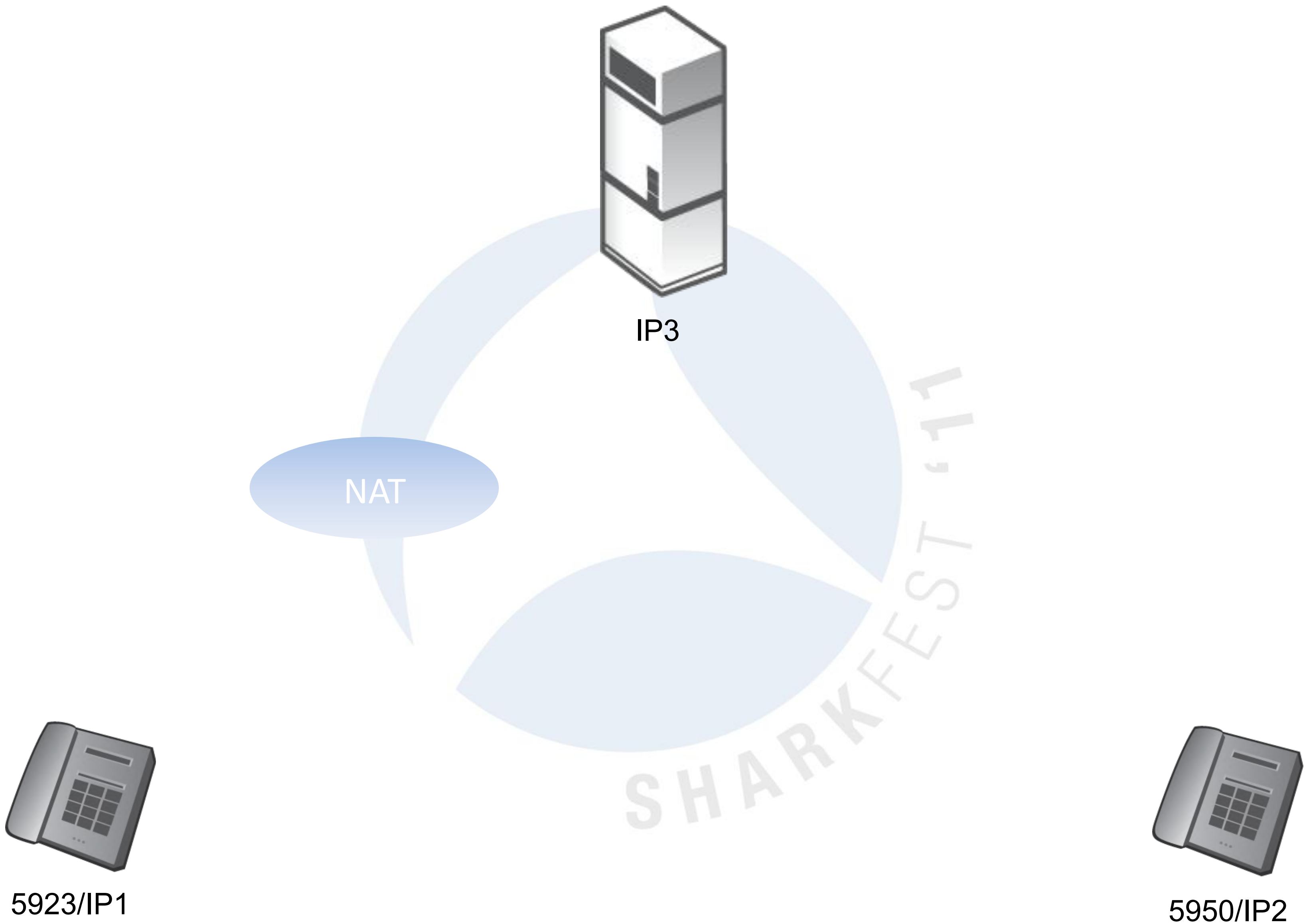


VoIP server as relay

– Cascade Pilot views (sip_ex_03_relay.pcap)
  - Transaction Analysis by VoIP Call

Calling from behind a NAT



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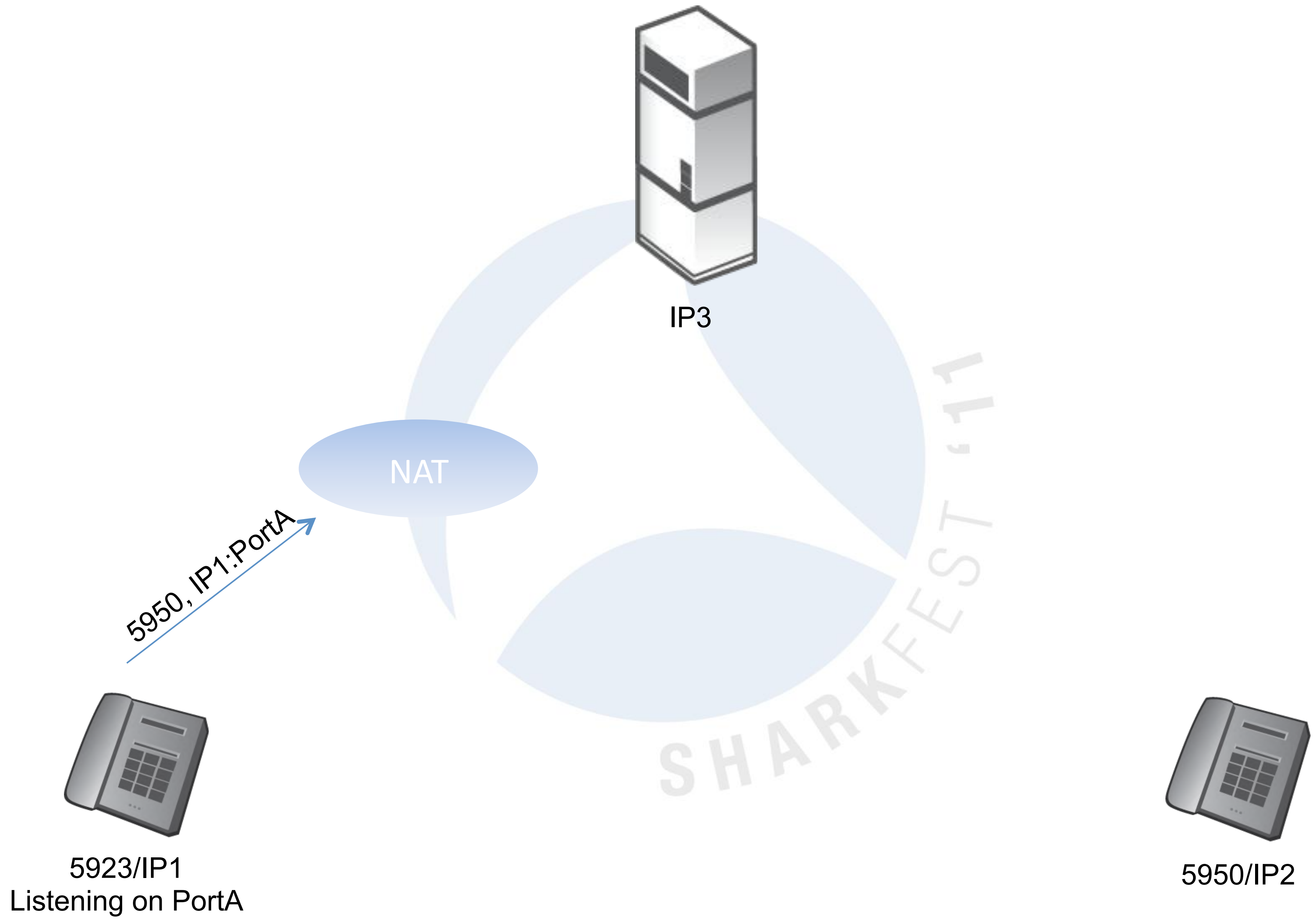

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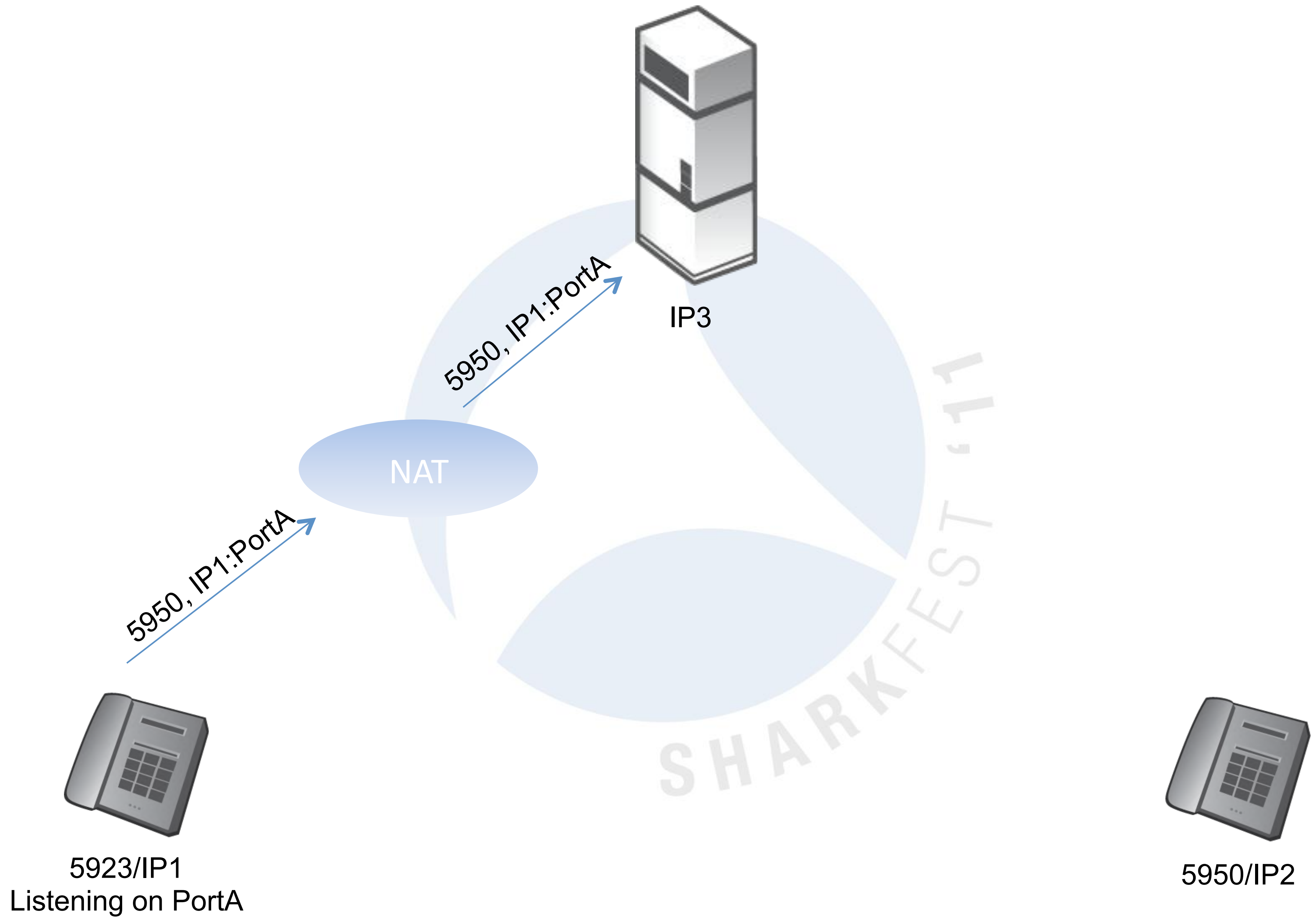

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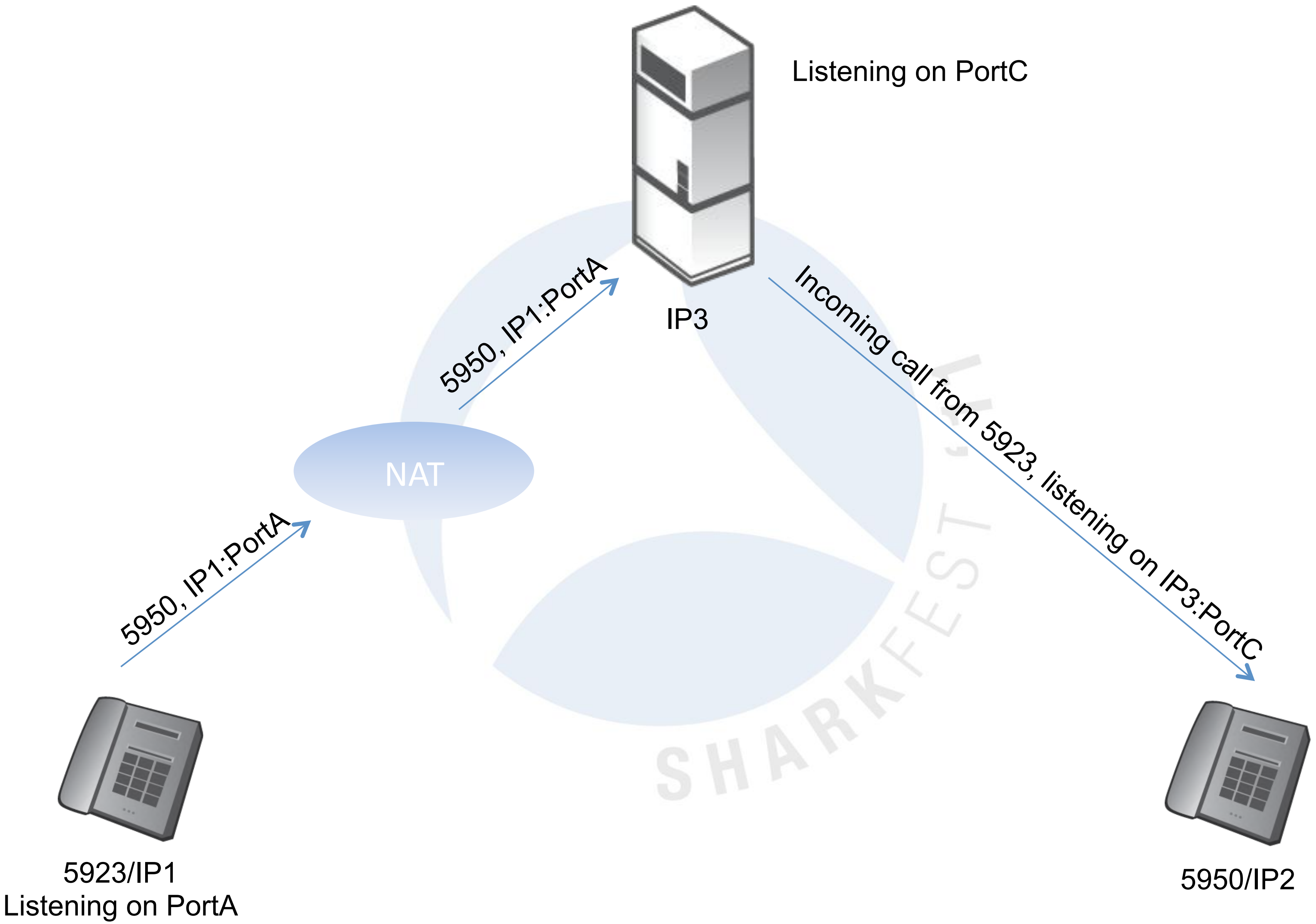

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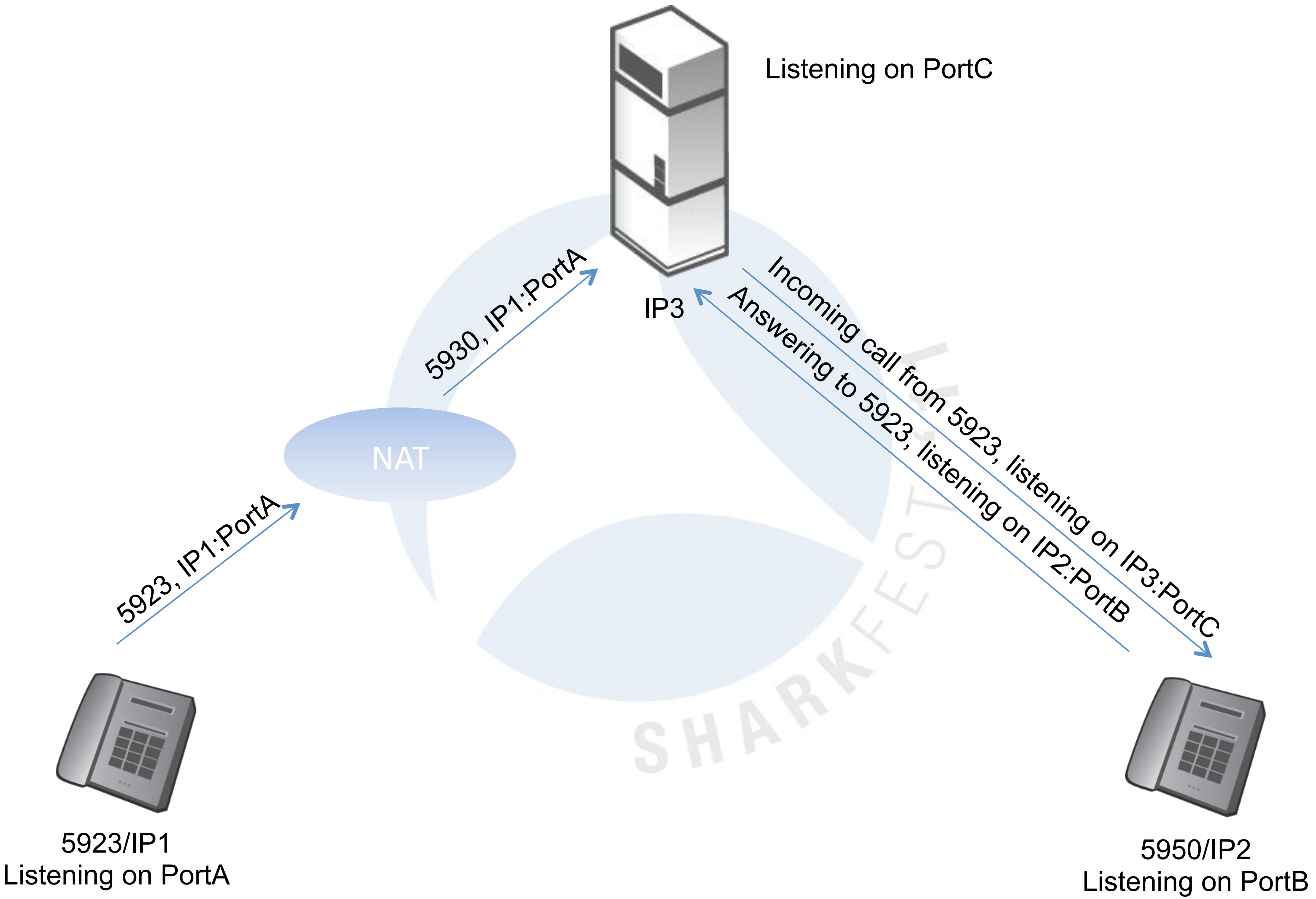

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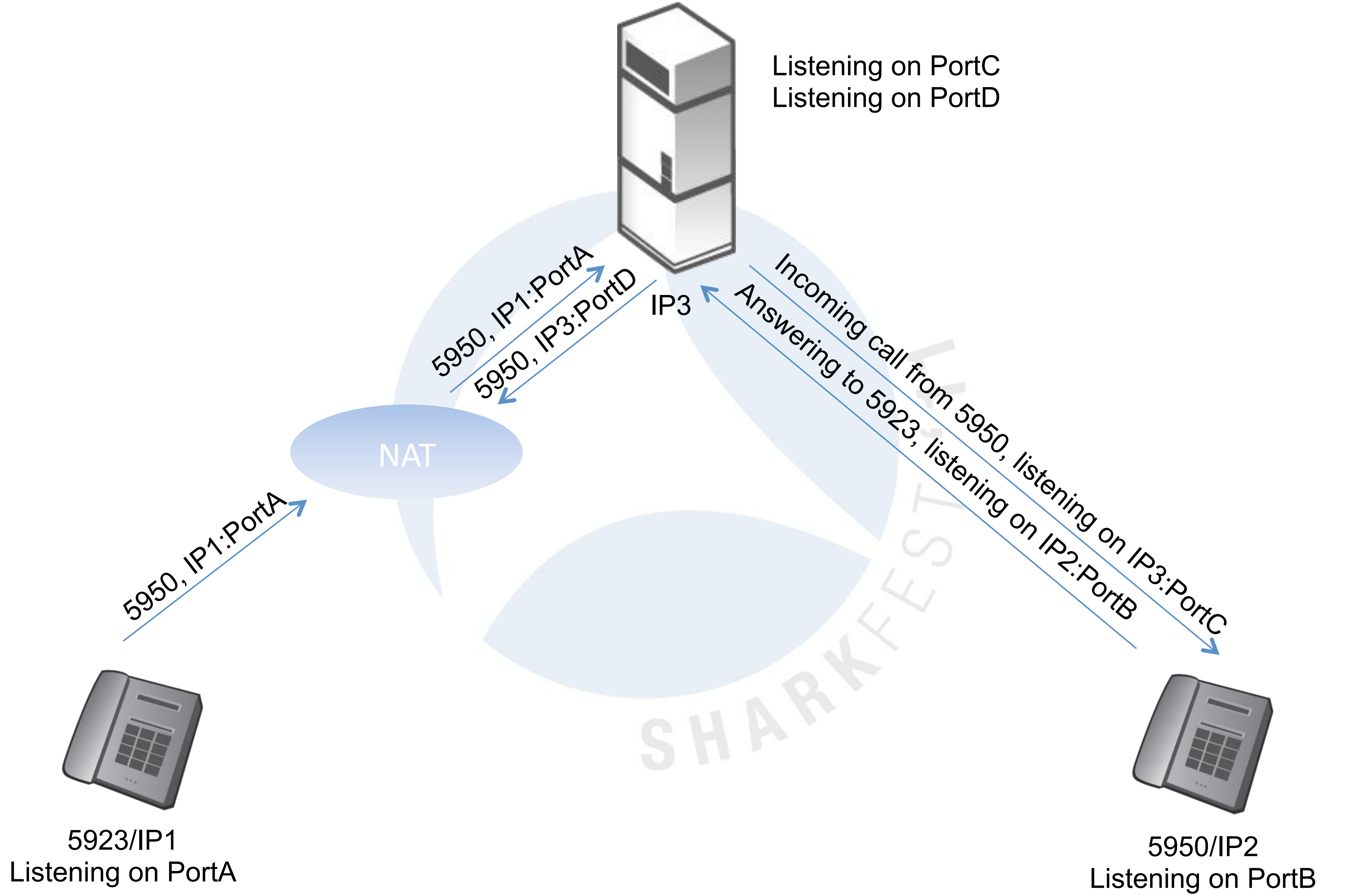

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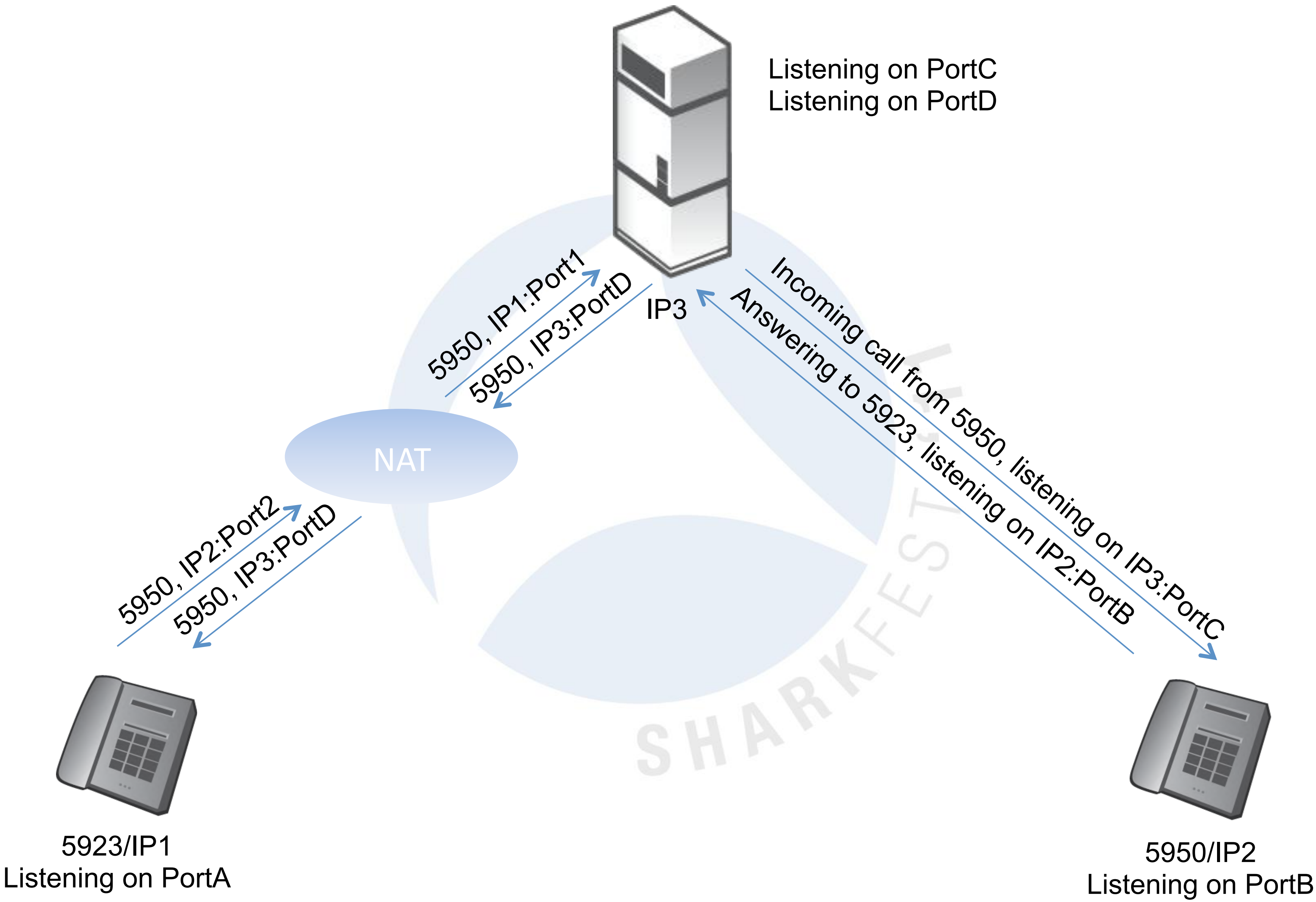

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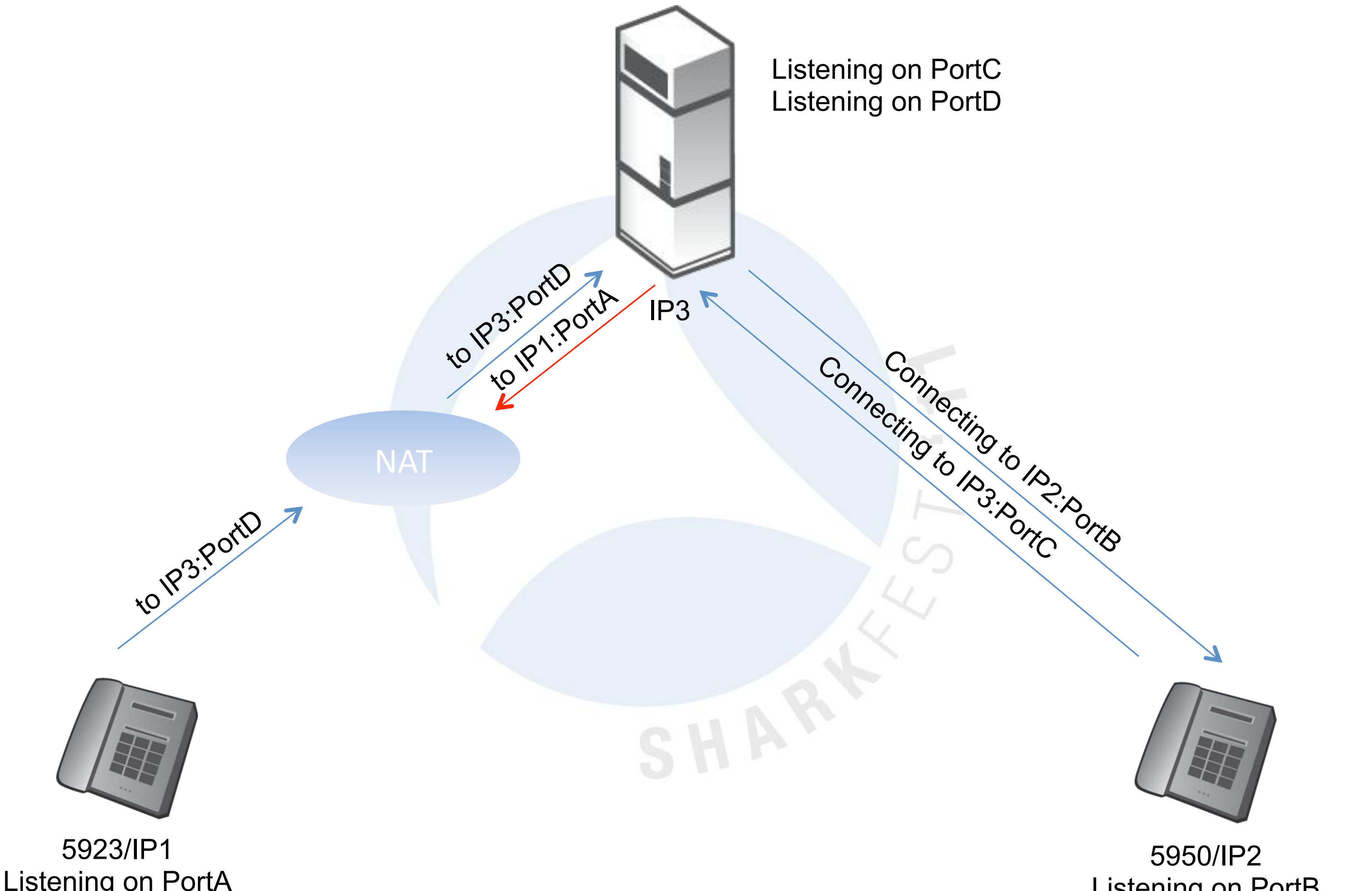

Calling from behind a NAT

- Cascade Pilot views
  - Transaction Analysis by VoIP Call
    - sip_ex_04_cace-voip-nat-behind-nat.pcap
    - sip_ex_04_cace-voip-nat-carl.pcap
    - sip_ex_04_cace-voip-nat-nina.pcap

VoIP Problems

- Signaling related problems
  - Control Messages/Errors
  - PDD
- Data related problems
  - Delay
  - Jitter
  - Packet Loss

VoIP Control Messages/Errors

- Recognizing for the different VoIP protocols:
  - Regular transaction messages
  - Misconfiguration errors
- SIP Response codes:
  - 1xx Provisional
  - 200 OK
  - 3xx Redirection
  - 4xx Request Failure
  - 5xx Server Failure
  - 6xx Global Failures

VoIP Control Messages/Errors

- Cascade Pilot views (sip_ex_01_sig_data.pcap)
  - SIP Massages Distribution

Post Dial Delay

- The Post Dialing Delay (PDD) is the time interval between the dialing of the last digit and the reception of either a ring tone or a busy signal
- The smaller PDD the better
- PDD <= 1s

Post Dial Delay

– Cascade Pilot views (sip_ex_01_sig_data.pcap)
  - VoIP Avg/Max/Min PDD Over Time
  - VoIP Avg PDD – Worst Sources/Callers
  - VoIP Avg PDD – Worst Destinations/Receivers

Voice Quality

- Packet Loss
- Latency
- Jitter
- R-Factor
- MOS

Packet Loss

– Data packet containing voice can be dropped
  - because of a congestion on Network
  - because of a limited buffer size on the receiver
– The Packet Loss in a voice stream affects the call quality
– Packet Loss <= 1 %

Packet Loss

- Cascade Pilot views (sip_ex_03._marcello.pcap)
  - VoIP Call Summary – Packet Loss
  - VoIP Avg/Max/Min Packet Loss Over Time
    - Send To Wireshark (RTP outside conversations)
  - VoIP Packet Loss – Worst Sources/Callers
  - VoIP Packet Loss – Worst Destinations/Receivers
  - Transaction Analysis by VoIP Call

Latency

- The Latency is the amount of time that takes for a data packet to get from a source to a destination
- For VoIP the Latency is the measure of the one-way delay between two nodes
- Latency <= 150 ms

Jitter

– The Jitter is the variation of the data packet arriving time, the variation of the data packet delay
– If correctly compensated with jitter buffer it does not affect too much a communication
– Jitter <= 20-30 ms

Jitter

– Cascade Pilot views (sip_ex_03._marcello.pcap)
  - VoIP Call Summary – Jitter
  - VoIP Jitter Distribution

R-Factor

- The R-Factor is a numeric method for computing and index of the quality of a voice stream (E-Model, ITU-T Recommendation G.107)
- The R-Factor takes into account
  - The cumulative effect of the used CODEC impairments
  - The Packet Loss and how it affects the used CODEC
  - Delay between two consecutive voice data packets
- The R-Factor ranges between 50 and 90
- R-Factor >= 80

MOS

- MOS (Mean Opinion Score) is a numeric value that provides an indication of the voice stream quality
- The MOS ranges between 1 (Impossible to communicate) to 5 (Perfect communication)
- MOS >= 4